SOUNDWIN
VoIP Solution Provider

TG1600/TG2400/TG3200 Series

VOIP Voice Gateway

USER MANUAL

Version: 1.1

Default Login Details

WAN IP Address: DHCP

LAN IP Address: http://192.168.1.1

User Name: root

Password: root

Preface 5
0.1 About this manual 5
0.2 Copyright 5
0.3 Trademark.....5
0.4 Safety Instructions.....5
0.5 Repair Warranty.....5
Introduction.....6
1.1 Overview.....6
1.2 Acronyms Table.....6
1.3 Introduction 7
1.4 Front panel LED description.....8
1.5 Product Specifications 9
Installation and setting.....11
2.1 packing contents 11
2.2 Hardware Installation 12
2.3 8 port/16 port/24 port/32 port Series module extension install.....13
2.4 Quick Start 14
Network setting 15
3.1 Internet.....15
3.1.1 Static IP 16
3.1.2 DHCP Mode.....17
3.1.3 PPPoE Mode.....18
3.1.4 PPTP with WAN (Static, DHCP, PPPoE) 19
3.1.5 L2TP with WAN (Static, DHCP, PPPoE).....20
3.2 LAN interface 21
3.2.1 User list.....21

3.3 Dynamic DDNS 22
3.4 QOS setting 23
3.5 NAT Mode / Bridge mode 23
SIP Setting.....24
4.1 SIP setting 24
4.1.1 Proxy server setting.....24
4.1.2 SIP Advance Setting 26
4.1.3 Codec 27
4.1.4 Line DC Feed Setting.....27
4.2 Telephone function 28
4.2.1 Call Forward Setting 28
4.2.2 Speed Dial 29
4.2.3 Dialing Plan 29
4.2.4 Do Not Disturb 30
4.2.5 Hotline Setting.....30
4.2.6 Caller ID display.....31
4.3 DSP 31
4.3.1 Volume Control 31
4.3.2 Current control loop 32
4.3.3 Flash Hook.....33
4.3.4 SLIC Setting 33
4.3.5 Polarity Reversal.....34
Advance setting 35
5.1 NAT Traversal.....35
5.2 Phone setting 35
5.2.1 Common 35
5.2.2 Region Tone Setting.....36

5.3 DSCP Setting....37
5.4 Voice Prompt....37
Security....38
6.1 Firewall....38
6.1.1 IP Filter....38
6.1.2 Mac Filter....38
6.1.3 URL Filter....39
6.1.4 Port Filter....40
6.2 DOS Prevention Setting....40
Maintenance....42
7.1 System....42
7.1.1 Account Setting....42
7.1.2 Time Setting....42
7.2 Tool....43
7.2.1 Backup / Restore....43
7.2.2 Firmware Upgrade....44
7.2.3 Ping testing....44
7.2.4 Restart....44
Installation structure....46

Preface

0.1 About this manual

This manual is intended to help users to the proper use of TG1600/TG2400/TG3200 series VOIP voice gateway.

The manual will be different based on the different firmware version, the latest content (specifications) subject to the original notice, if any content in this document is supposed to change without notice.

0.2 Copyright

All rights reserved for the 2011 Telephony Corporation. All rights reserved. The information contained in this publication is protected by copyright. Without written permission, no part of the copy, transmitted, transcribed, or translated into any language, will be stored in a retrieval system, ownership of copyright owners.

0.3 Trademark

Appear in this manual product and company names may be related to the company's registered trademark that appears in this manual product and company names used only for identification or explanation.

0.4 Safety Instructions

Use only the required power supply voltage. Power input: AC 100 - 240V, 50 - 60 Hz. To reduce the risk of electric shock, do not disassemble this product. Opening or removing components could be dangerous. Incorrect assembly may result in abnormal or damaged product, please refer to the instruction manual before use.

Do not place any objects or liquid into the device, may cause electric shock, abnormal or damaged products. Products will not produce any liquid. If the product is found around the liquid leaks, identify and find the appropriate service personnel.

Use the unshielded twisted pair (UTP) Ethernet cable connection RJ - 45 port.

0.5 Repair Warranty

Users buy TG1600/TG2400/TG3200 series VOIP voice gateway products from the dealer have one year warranty. Be sure to obtain proof of purchase to the dealer and keep it. During the warranty period, if the product has any abnormalities, failure resulting in the not function proper, we will evaluate the repair fee or replacement of defective products or parts without charge. If the product is modified, abuse, tampering, destruction by improper use, or under abnormal operating conditions, such behavior is not apply the warranty policy. This warranty does not include other ancillary products authorized software vendor or accessories.

1

Introduction

TG1600/TG2400/TG3200 Series VoIP Gateway (VOIP Gateway) and Call Manager (Call Manage) include low-to high-end Internet telephony overall solution. This article describes how to use the Voice Gateway (VOIP Gateway) and Call Manager (Call Manage).

1.1 Overview

- Internet Telephony Gateway (VOIP Gateway) is a communications device that can be integrated analog phone, PSTN, or PBX trunk card/extension card.
- By Modular hardware design TG1600/TG2400/TG3200 series makes different combination of the telephone interface. By modular installation, this revolutionary compact voice over IP (VoIP) gateway could be easily configured as 16/24/32 channels high density FXS/FXO VoIP Gateway which provides voice connectivity over the IP network and to the Public Switched Telephone Network (PSTN).
- Provide a direct analog port. For computer modulation decoders, fax machines, analog phones and other devices, the analog port is necessary.
- Support for standard Internet services such as NAT, Dynamic DNS (DDNS), Quality of Service (QoS), Port Filtering, IP Filtering.

1.2 Acronyms Table

Acronym:	Full Name:	Acronym:	Full Name:
API	Application Interface	ACI	Audio CODEC Interface
ADC	Analog to Digital Converter	CODEC	Coder / Decoder
DAC	Digital to Analog Converter	DC	Direct Current
DDNS	Dynamic Domain Name System	DHCP	Dynamic Host Configuration Protocol
DMZ	Demilitarized Zone	DNS	Domain Name System
DTMF	Dual Tone Multi Frequency	FXO	Foreign Exchange Office
FXS	Foreign Exchange Station	GMT	Greenwich Mean Time
IP	Internet Protocol	IPSec	Internet Protocol Security
L2TP	The Layer 2 Tunnel Protocol	LAN	Local Area Network
WAN	Wide Area Network	MAC	Media Access Control
MII	Media Independent Interface	NAT	Network Address Translation

NTP	Network Time Protocol	PPTP	Point-to-Point Tunneling Protocol
RTP	Real-Time Transport Protocol	RTCP	Real-Time Transport Control Protocol (also known as RTP control protocol)
SIP	Session Initiation Protocol	SLIC	Subscriber Line Interface Circuit
STUN	Simple Traversal of UDP through NATs	URI	Uniform Resource Identifier
TCP	Transmission Control Protocol	UDP	User Gateway gram Protocol
UPnP	Universal Plug and Play	VoIP	Voice Over Internet Protocol

1.3 Introduction

- Support SIP VoIP protocol.
- TG0800 series: allow 8 SIP calls
  TG1600 series: allow 16 SIP calls
  TG2400 series: allow 24 SIP calls
  TG3200 series: allow 32 SIP calls

Model	FXS Port	FXO port	WAN Port	LAN Port	RJ-11 port	SIP
TG0800	8		1	1	8	√
TG0804	4	4	1	1	8	√
TG0808		8	1	1	8	√
TG1600	16		1	1	16	√
TG1608	8	8	1	1	16	√
TG1616		16	1	1	16	√
TG2400	24		1	1	24	√
TG2412	12	12	1	1	24	√
TG2424		24	1	1	24	√
TG3200	32		1	1	32	√
TG3216	16	16	1	1	32	√
TG3232		32	1	1	32	√

Others:

Model	Description	
TG-Host	1 WAN port, 1 LAN port with 1U Rack	

TM800	8FXS Module	
TM804	4FXS + 4FXO Module	
TM808	8FXO Module	

1.4 Front panel LED description

LED	Status	Indication
Power	On	GW is power on
	Off	GW is power off
RUN	On	GW connection established
	Flashing	Data traffic on cable network
	Off	Waiting for GW connection
USB	On	USB connected
	Off	Wait for USB connected
LAN	On	LAN is connected successfully
	Flashing	Data is transmitting
	Off	Ethernet not connected to PC
WAN	On	GW network connection established
	Flashing	Data traffic on cable network
	Off	Waiting for network connection
FXS	On	Telephone Set is Off-Hook
	Flashing	Ring Indication
	Off	Telephone Set is On-Hook

FXO	On Off	Line is Off-hook Line is On-hook

1.5 Product Specifications

Telephone Feature
Voice Codec: G.711 / G.729 / G.723 / G.726
Battery Reverse Generation
Off-Hook Loop Current Configuration
Ring Voltage Configuration
Internet Failed / Register Failed by pass PSTN
DTMF Relay support : In-Band / RFC-2833 / SIP-Info
Voice channels status display: Display each port status like as ON-Hook, OFF-Hook, calling number.
FAX support : T.38 FAX / FAX by G.711 pass-through
AC termination Impedance : 600/900 OHM and complex impedance
Answer Supervision for polarity reverse detection
IP Specifications:
SIP (RFC 3261), SDP (RFC 2327), STUN (RFC3489) ,RTP Payload for DTMF Digits (RFC2833) SIP Session Timers (RFC 4028), DNS SRV (RFC 2782) , Outbound Proxy Support , SIP REFER method (RFC 3515), Early Media and Ringing Tone Generation (RFC 3960), Message Waiting Indicator (RFC 3842)
WAN: PPPoE client, DHCP client, Fix IP Address, PPTP
Support PPTP/L2TP VPN Client
NAT Functions
DSCP configuration
QoS : DSCP / VLAN Tag
Network and Security specifications
Port Filter
IP Filter
MAC Filter
Call Feature
Adjustable Volume
VAD / Dynamic Jitter Buffer
Caller ID Generation : DTMF CID / FSK CID / NTT CID
Peer-to-Peer Call by Dial Plan
Different Country Tone Table Configuration
Configuration and Management
Web-Based Graphical UI Configuration
HTTP Firmware Upgrade

SNMP v1/v2 (Customization Project)
IVR
Provision
TR069/TR098/TR104 (Project Base)
Auto-Provision by HTTP/TFTP/FTP (Project Base)
General Specification
AC power: AC100V-240V, DC12V/2.5A, 50/60 Hz
Temperature: 0℃ ~ 40℃ (Operation)
Humidity: up to 90% non-condensing
Emission: FCC Part 15 Class B, CE Mark, RoHS Compliant
Dimension : 435x255x45 mm
Weight: 3000 g

2

Installation and setting

2.1 packing contents

Please check the closed packing of products and accessories before installation. The following are the default contents of the product. The contents of the actual product may be varied according to different versions.

TG1600/TG2400/TG3200 series VOIP Gateway packing contents

TG-Host

TM800 module

TM804 module

TM808 module

Adapter

RG-45 cable

User manual

TG0800 series packing contents	
TG-Host	X1
TM800/TM804/TM808 module	X1
RJ-45cable	X1
AC adaptor (2.5 A)	X1

CD-Rom (user manual)	X1

TG1600 series packing contents	
TG-Host	X1
TM800/TM804/TM808 module	X2
RJ-45cable	X1
AC adaptor (2.5 A)	X1
CD-Rom (user manual)	X1

TG2400 series packing contents	
TG-Host	X1
TM800/TM804/TM808 module	X3
RJ-45cable	X1
AC adaptor (2.5 A)	X1
CD-Rom (user manual)	X1

TG3200 series packing contents	
TG-Host	X1
TM800/TM804/TM808 module	X4
RJ-45cable	X1
AC adaptor (2.5 A)	X1
CD-Rom (user manual)	X1

2.2 Hardware Installation

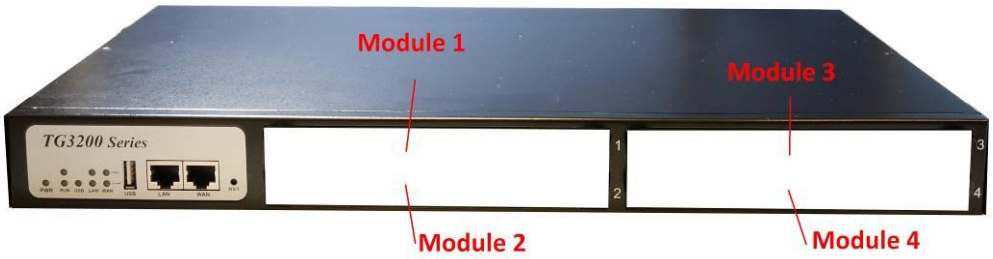
item	Connect interface	Function
1	FXS	FXS port connected to phone or pbx outbound card.
2	FXO	FXO port connected to PBX inbound card or CO line Back up for life line support, it's used for PSTN call if VOIP Call fail.
3	WAN	Connected to internet. The WAN port is able to connect the gateway and modem (such as cable modem, ADSL modem) to internet.

4	LAN	Connected to PC or switch /hub. Let your PC or Switch /hub connected gateway to create LAN environment.
5	RESET bottom	Push the reset bottom till 3 sec. Gateway will be set to default
6	AC power(DC in 12V)	Power input

2.3 8 port/16 port/24 port/32 port Series module extension install

TG0800/TG1600/TG2400/TG3200 series, according to different module board (TM800/TM804/TM808) and TG-Host device combine to 8/16/24/32 port high level gateway.

- TG-Host: 1 WAN port, 1 LAN port with 1U Rack
- TM800: 8FXS module
- TM804: 4FXS+4FXO module
- TM808: 8FXO module



The models are as follows:

TG-Host	Module_1	Module_2	Module_3	Module_4	Model
√	TM800: 8 FXS				TG0800: 8FXS
√	TM804: 4FXS+4FXO				TG0804: 4FXS+4FXO
√	TM808: 8FXO				TG0808: 8FXO
√	TM800: 8 FXS	TM800: 8 FXS			TG1600: 16FXS
√	TM804: 4FXS+4FXO	TM804: 4FXS+4FXO			TG1608: 8FXS+8FXO
√	TM808: 8FXO	TM808: 8FXO			TG1616: 16FXO
√	TM800: 8 FXS	TM800: 8 FXS	TM800: 8 FXS		TG2400: 24FXS
√	TM804: 4FXS+4FXO	TM804: 4FXS+4FXO	TM804: 4FXS+4FXO		TG2412: 12FXS+12FXO
√	TM808: 8FXO	TM808: 8FXO	TM808: 8FXO		TG2424: 24FXS

√	TM800: 8 FXS	TM800: 8 FXS	TM800: 8 FXS	TM800: 8 FXS	TG3200: 32FXS
√	TM804: 4FXS+4FXO	TM804: 4FXS+4FXO	TM804: 4FXS+4FXO	TM804: 4FXS+4FXO	TG3216: 16FXS+16FXO
√	TM808: 8FXO	TM808: 8FXO	TM808: 8FXO	TM808: 8FXO	TG3232: 32FXO

2.4 Quick Start

How to set up TG1600/TG2400/TG3200 series VOIP gateway?

1. connect your PC or NB to LAN port of TG1600/TG2400/TG3200 series voip gateway
2. key in http://192.168.1.1 on web browser (IE / Firefox)
3. enter user name and password

User name: admin password: admin

4. Enter wizard mode
   1. Connection Setup
   2. VOIP Setup

TG1600/TG2400/TG3200 series voip gateway internet setting

1. Time zone setting
2. LAN IP, subnet mask IP setting (please keep default setting if not necessary to change the IP))
3. WAN internet setting

   WAN connection type, please choose your networking type

   Static IP (fixedIP, IP information provided by ISP)

   DHCP Clinet (distributing IP automatically)

   PPPoE (account / password provided by ISP)

5. click “end” and finish the internet setting, then, go to “VOIP Setup” to configure the SIP account.
6. connect TG1600/TG2400/TG3200 series voip gateway to your internet environment (connect RJ45 cable to WAN port of TG1600/TG2400/TG3200 series voip gateway).
7. The device will set up voip information automatically if devices setting finished
8. Start to call voip call

3

Network setting

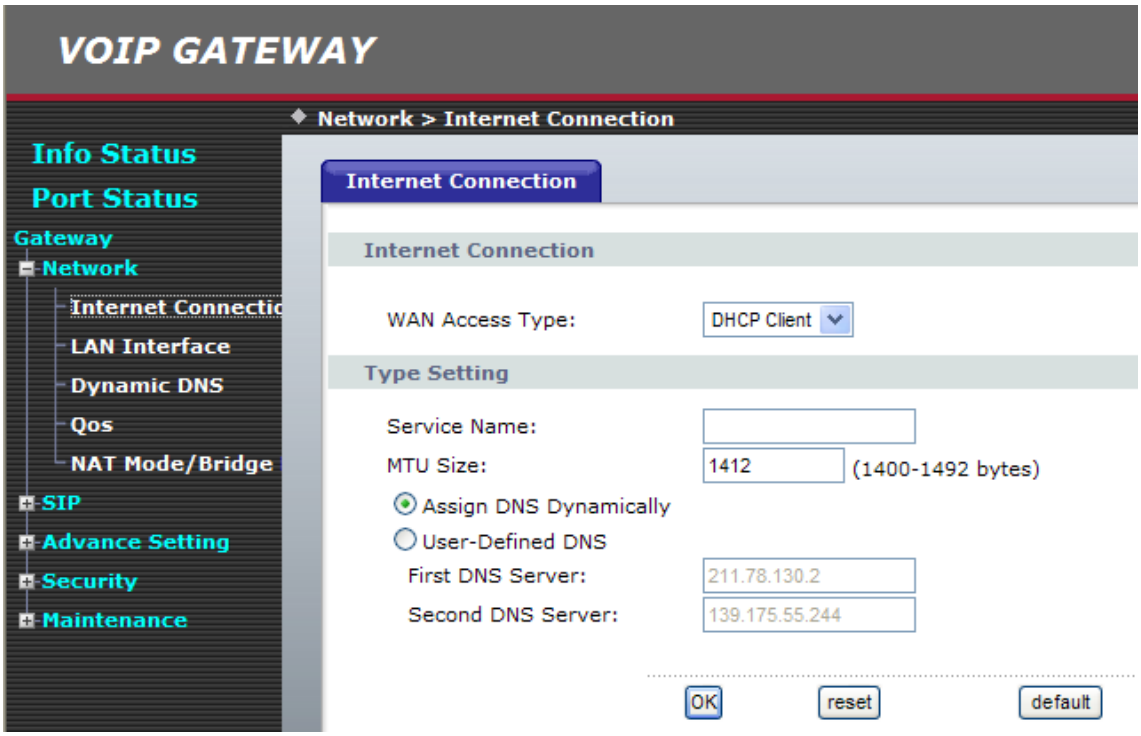
3.1 Internet

WAN (Wide Area Network) is a network connection connecting one or more LANs together over some distance. For example, the means of connecting two office buildings separated by several kilometers would be referred to as a WAN connection. The size of a WAN and the number of distinct LANs connected to a WAN is not limited by any definition. Therefore, the Internet may be called a WAN.

WAN Settings are settings that are used to connect to your ISP (Internet Service Provider). The WAN settings are provided to you by your ISP and often times referred to as "public settings". Please select the appropriate option for your specific ISP.

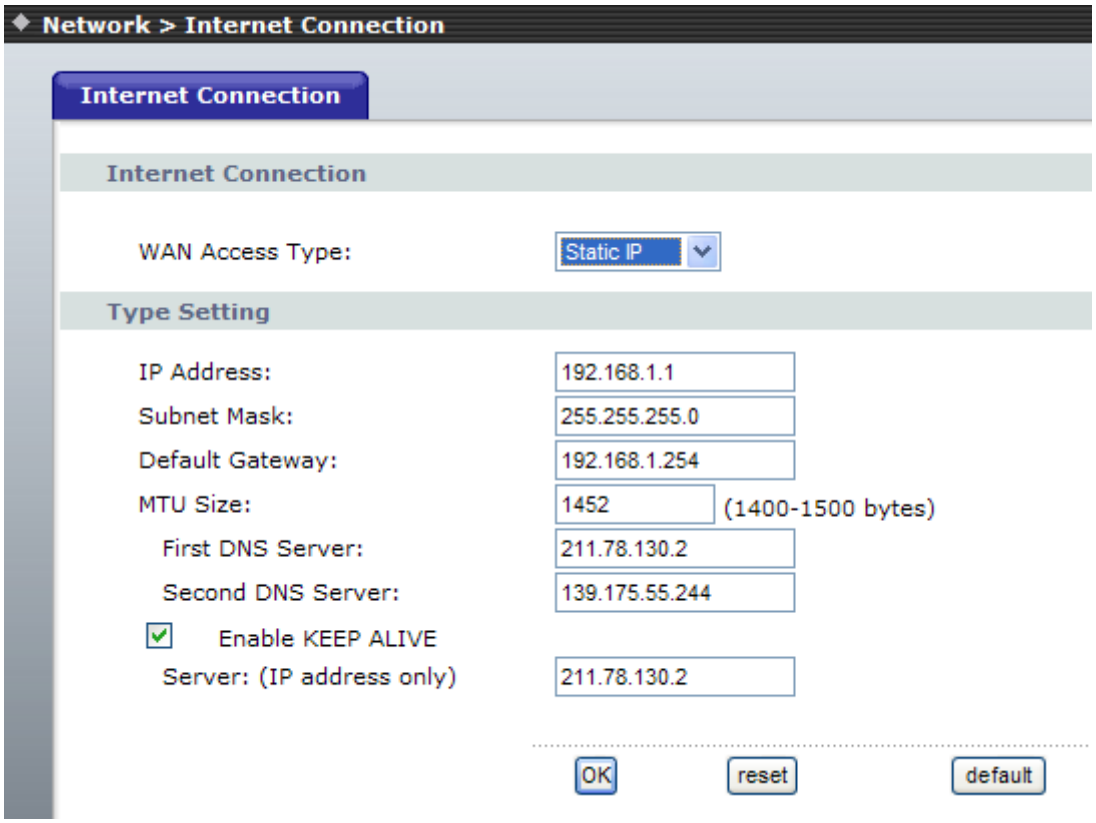
For most users, Internet access is the primary application. TG1600/TG2400/TG3200 series voip gateways support the WAN interface for internet access and remote access. The following sections will explain more details of WAN Port Internet access and broadband access setup. When you click “WAN Setting”, the following setup page will be shown. Three methods are available for Internet Access.

1. Static IP
2. DHCP
3. PPPoE



3.1.1 Static IP

If you are a user with static IP address, please enter the IP address, subnet mask, default gateway and DNS servers, which are provided by your ISP (Internet Service Provider). Each IP address must be entered in the field of IP in the appropriate forms, which are four IP octets separated by points (XXXX). Router will not be accepted if IP address not like in this format. For example: 168.95.1.2



□IP Address: Check with your ISP provider.

□Subnet Mask: Check with your ISP provider.

□Default Gateway: Check with your ISP provider.

□MTU size: MTU stands for Maximum Transmission Unit, the largest physical packet size, measured in bytes that a network can transmit. Any messages larger than the MTU are divided into smaller packets before being sent.

The key is to be deciding how big your bandwidth pipe is and select the best MTU for your configuration. For example, you have a 33.6 modem, you use a MTU of 576, and if you have a larger pipe you may want to try 1500.

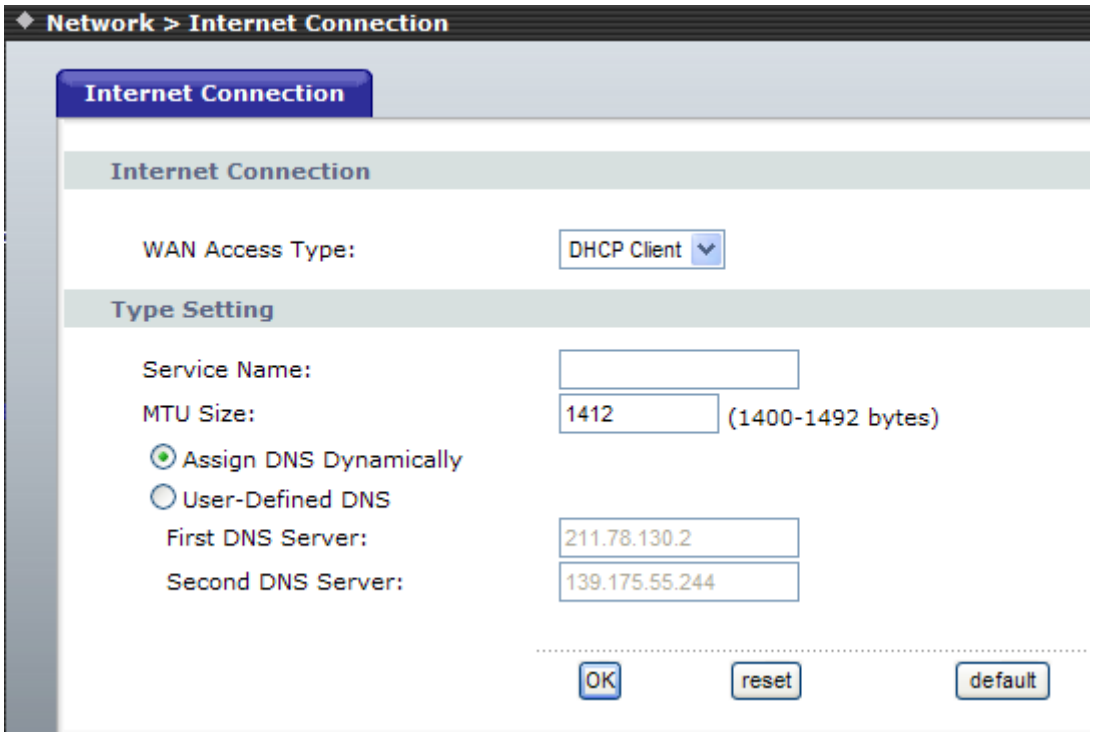
□DNS server: enter DNS server (suggest default setting)

□Enable Keep Alive

3.1.2 DHCP Mode

Dynamic Host Configuration Protocol (DHCP), Dynamic IP (Get WAN IP Address automatically). If you are connected to the Internet through a Cable modem line, then a dynamic IP will be assigned.

Note: WAN port gets the IP Address, Subnet Mask and default gateway IP address automatically, if DHCP client is successful.



□Name of service: it may be set by your own or blank

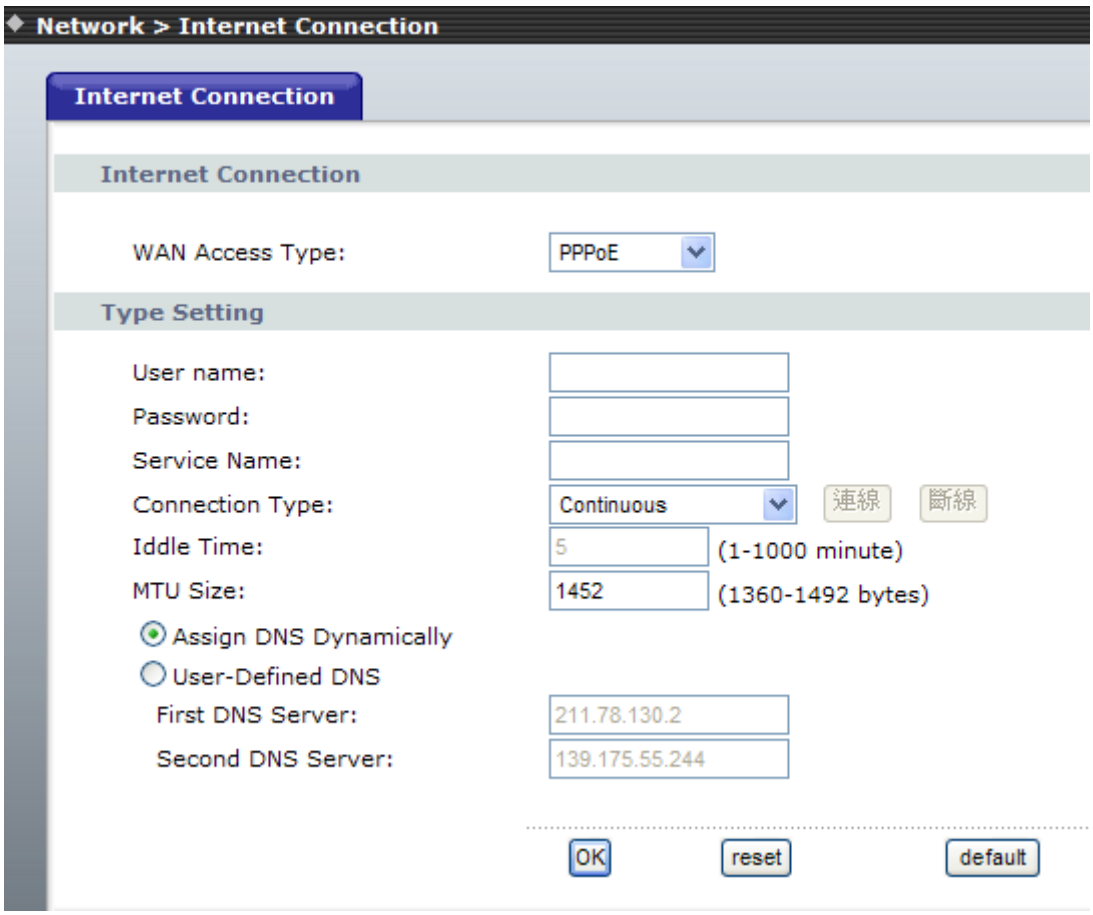
□MTU size: enter MTU (Maximum transmission unit) size (suggest set to default)

□Assign DNS Dynamically: get DNS IP address WAN interface automatically (suggest use it)

□First DNA server, Second DNS server: Manually enter the domain name server WAN interface

3.1.3 PPPoE Mode

Point-to-Point Protocol over Ethernet (PPPoE). Some ISPs provide DSL-based services and use PPPoE to establish communication link with end-users. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. If they do, you need to make sure the following items, PPPoE User name: Enter username provided by your ISP. PPPoE Password: Enter password provided by your ISP.



□USER name: the user name provided by your ISP

□user password: enter password provided by your ISP

□Name of service: you may set up the name of service or blank

□Type of connection:

Continuous: will keep trying to connect internet while disconnect

Connect on demand: will be set according to the idle time when connection

□Manual: connect or disconnect by manual

□Idle time: when the connection type is: Connect on demand, can set up the interval time to reconnect internet

□MTU size: enter MTU (max transmission unit) size (suggest set to default)

□Server 1,2: Manually enter the domain name server WAN interface

3.1.4 PPTP with WAN (Static, DHCP, PPPoE)

Some ISPs provide DSL-based service and use PPTP with Static, DHCP or PPPoE to establish communication link with end-users.

□PPTP Server IP Address: check with your VPN /PPTP Server provider

□PPTP User Name: PPTP Dial-in account

□PPTP Password: PPTP Dial-in Password

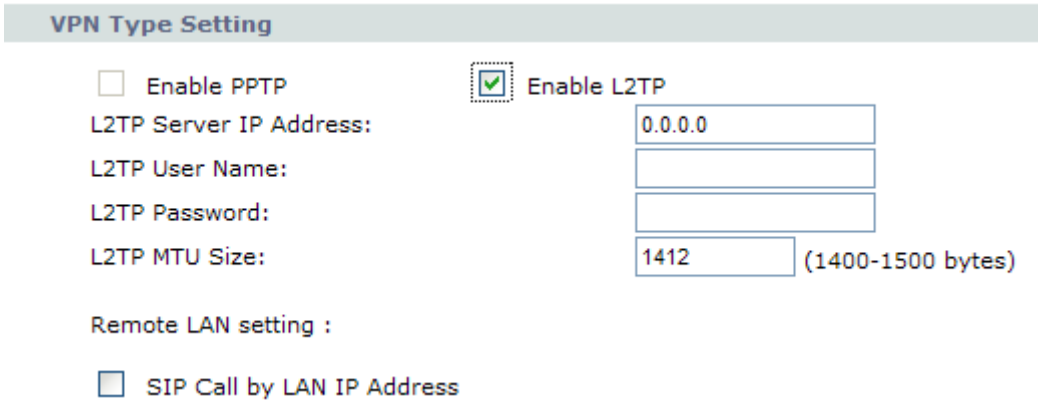
□PPTP MTU Size

□Request MPPE Encryption

□Remote LAN setting: select SIP Call by LAN IP Address or not

3.1.5 L2TP with WAN (Static, DHCP, PPPoE)

Some ISPs provide DSL-based service and use L2TP with Static, DHCP or PPPoE to establish communication link with end-users.



□L2TP Server IP Address: check with your VPN /L2TP Server provider

□L2TP User Name: L2TP Dial-in account

□L2TP Password: L2TP Dial-in Password

□L2TP MTU Size

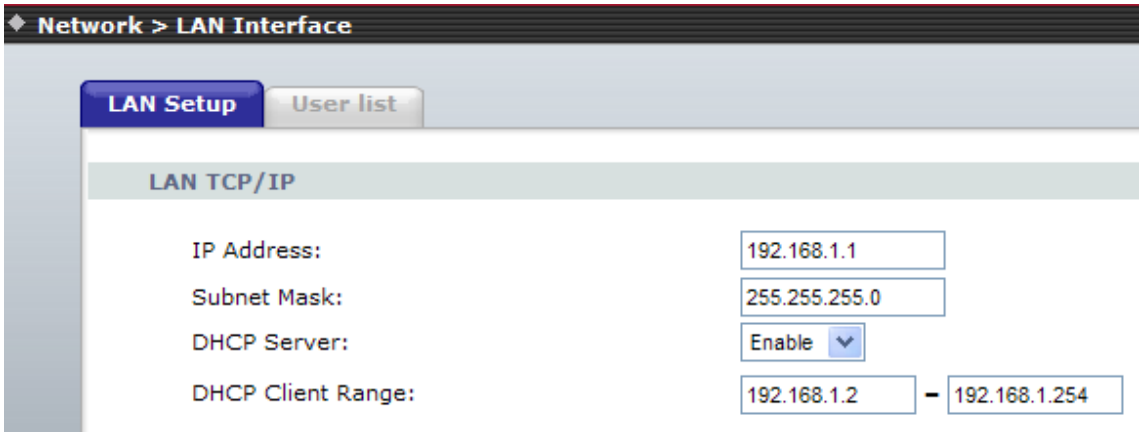
□Request MPPE Encryption

□Remote LAN setting: select SIP Call by LAN IP Address or not

3.2 LAN interface

The IP settings of the LAN (Local Area Network) interface for the device. These settings may be referred to as "private settings". You may change the LAN IP address if needed. The LAN IP address is private to your internal network and cannot be seen on the Internet. The default IP address is 192.168.1.1 with a subnet mask of 255.255.255.0.

LAN is a network of computers or other devices that are in relatively close range of each other. For example, devices in a house or office building would be considered part of a local area network



□LAN IP Address: Assign the IP address of LAN server, default is 192.168.1.1

□Subnet Mask: Select a subnet mask from the pull-down menu, default is 255.255.255.0.

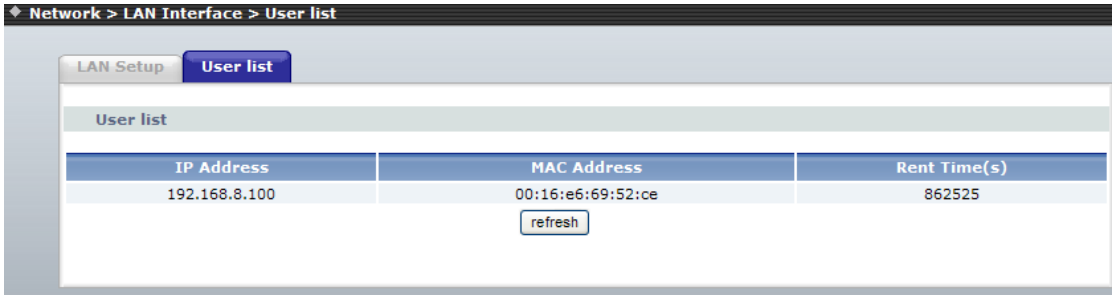
□DHCP Server Setting: DHCP stands for Dynamic Host Control Protocol. The DHCP server gives out IP addresses when a device is starting up and request an IP address to be logged on to the network. The device must be set as a DHCP client to "Obtain the IP address automatically". By default, the DHCP Server is enabled in the unit. The DHCP address pool contains the range of the IP address that will automatically be assigned to the clients on the network.

When you enable the DHCP server,

□DHCP Client Range: Enter the starting IP address for the DHCP server's IP assignment and the ending IP address for the DHCP server's IP assignment.

3.2.1 User list

The client computer is connected to the unit will display their information table in the DHCP client list. The table displays the IP address, MAC address, and the lease time, DHCP lease statistics for each time the client computer. Click Refresh to update the screen display



3.3 Dynamic DDNS

Maintenance of dynamic database of domain names, and the corresponding "Internet Protocol"(IP) address

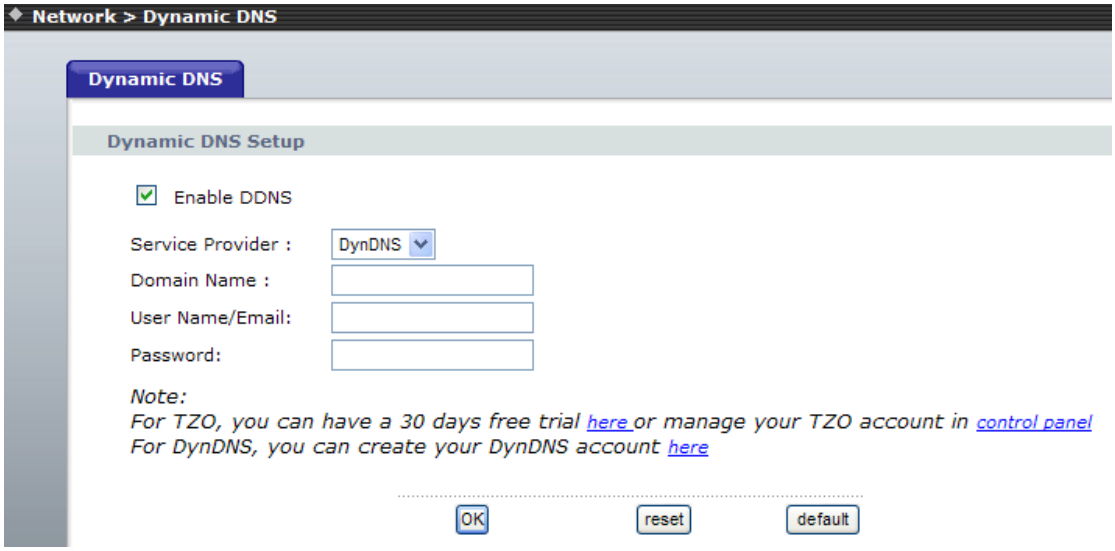
DNS: is a core Internet services, as domain names and IP addresses can be mapped to a distributed database, can make people more convenient access to the Internet, without having to remember that the machine can be directly read a string of IP

How to use dynamic domain name service

From this web site you can register a new DDNS service account:

http://www.dyndns.com/newacct

Note that if you are using a fixed IP address, do not set the gateway dynamic domain name. DDNS and use both a fixed IP, DynDNS DDNS service will stop your DDNS service.



□Enable DDNS: Enable/Disable the DDNS service, default setting is Disable.

□DDNS Server Type: Support two types of DDNS, DynDns.org or TZO.net

□Domain Name: The domain which you register in DynDns.org or TZO.net website.

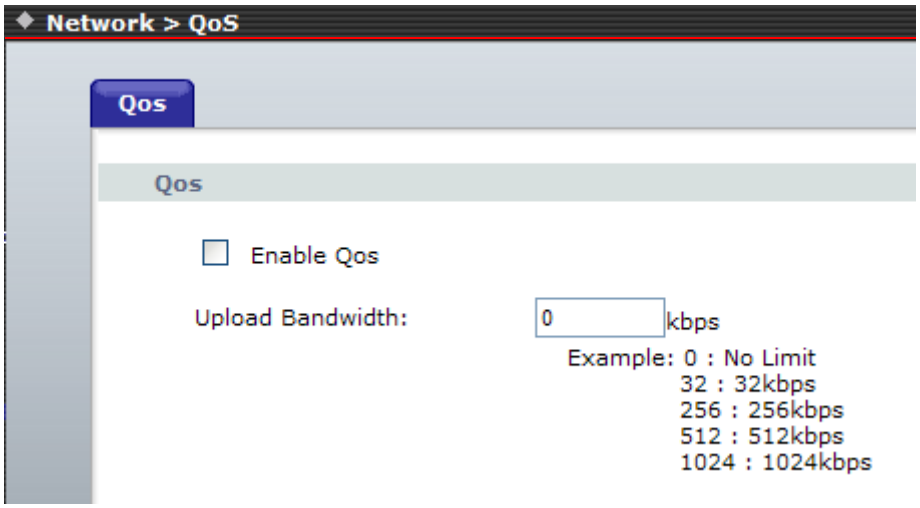
□DDNS Username : The username which you register in DynDns.org or TZO.net website.

□DDNS Password: The password which you register in DynDns.org or TZO.net website.

3.4 QOS setting

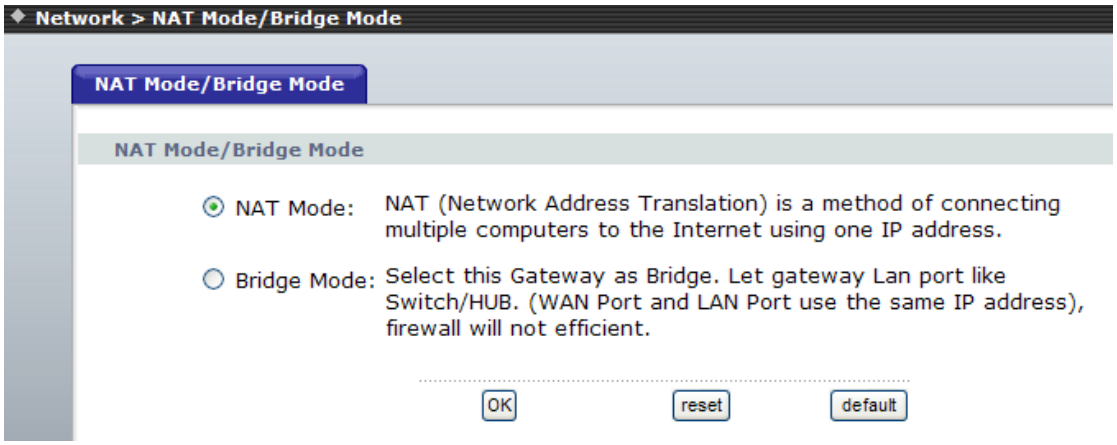
When using a VoIP call, in order to ensure the voice bandwidth, the other connection will automatically reduce the transmission capacity

You can set your Qos upload speed



□Enable QoS: Enable/Disable the QoS service

3.5 NAT Mode / Bridge mode


Network > NAT Mode/Bridge Mode
NAT Mode/Bridge Mode
NAT Mode/Bridge Mode
NAT Mode:
NAT (Network Address Translation) is a method of connecting multiple computers to the Internet using one IP address.
Bridge Mode:
Select this Gateway as Bridge. Let gateway Lan port like Switch/HUB. (WAN Port and LAN Port use the same IP address), firewall will not efficient.
OK
reset
default

□NAT Mode: This mode allows LAN users to share a single IP address and Internet connection.

□Bridge Mode: This mode allows the Internet interface and local area network interface bridging, Notice: the bridge mode failure once you start NAT and firewall

4

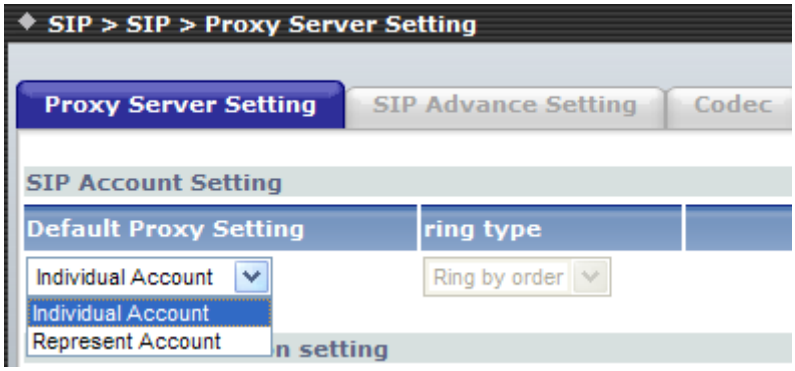
SIP Setting

4.1 SIP setting

SIP is a request-response protocol, dealing with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol. SIP determines the end system to be used for the session, the communication media and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination.

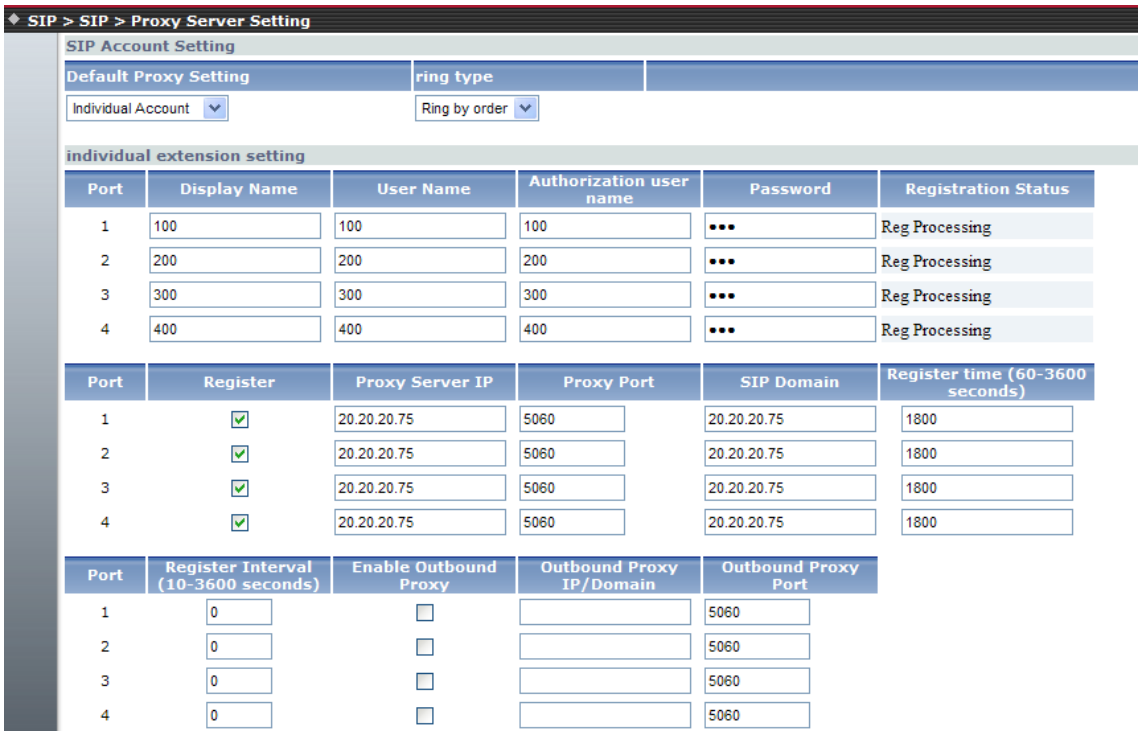
4.1.1 Proxy server setting

Support “Individual Account” or “Represent Account” setting.



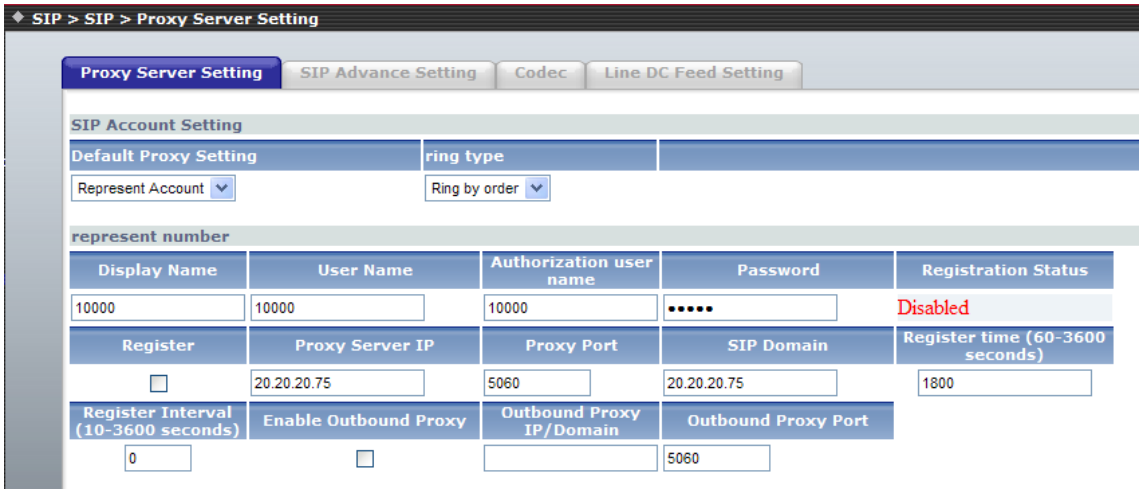
□Individual Account:

Can register multiple accounts with different phone use, respectively, accounting can be independent. Need to apply account from System operators



□Represent Account:

Representative number is used for PBX integration, between the objects can only remember a representative number is enough, but the account can't be independent. Need to apply account from System operators

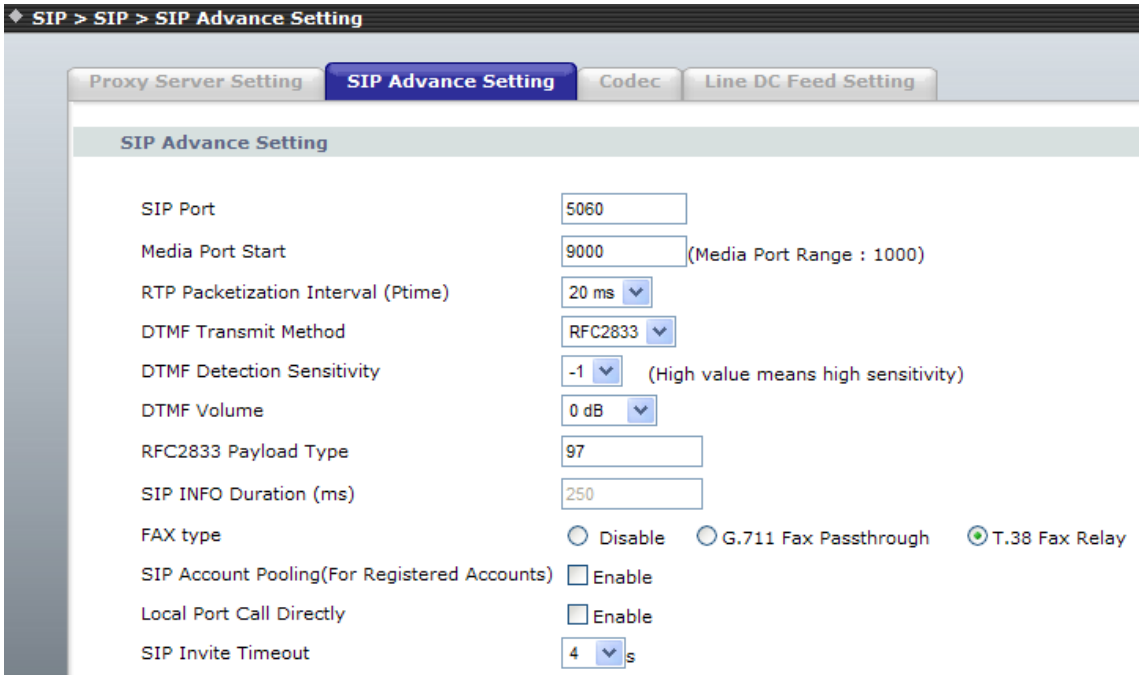


□Ring type:

Ring by order: Each time sequence starting from the line 1 is not occupied by line looking for ring

Round Robin: Turn calls increased each cycle, ringing every user

4.1.2 SIP Advance Setting



□SIP port number: Local SIP port number setting defaults:5060

□Media Port Start: The starting range of RTP port. Port number for initial of sending RTP packet, default setting is 9000.

□RTP Packetization interval (Ptime):

Value less: Increase network load and improve the sound quality

Value gain: Reduce the network load, reduce the sound quality.

□DTMF Transmit method: Select DTMF transmission format RFC2833, Sip info, In band. Default is RFC2833

□DTMF detection Sensitivity:

Value gain: Sensitivity enhancement, more digit issue may happen, suitable for enlarge the item when dial short length number

Value less: Lower sensitivity, less digit issue may happen, suitable for enlarge the item when dial the long length number.

Default is -1

□DTMF volume: Sometimes no action when dial, you may adjust the function.

For example: when you make call to a company, you get a IVR, but no action when dial the extension number, you may increase the value. Default is 0dB

□RFC2833 Payload Type: Sending the DTMF tone as a RTP payload signal. The RFC2833 signaling

□SIP INFO Duration (ms): Modify SIP INFO duration time.

□FAX type:

G711 fax pass through (not compressed by using a 64Kbps channel to transmit voice signals)

T.38 fax relay (fax encoded by IP protocol)

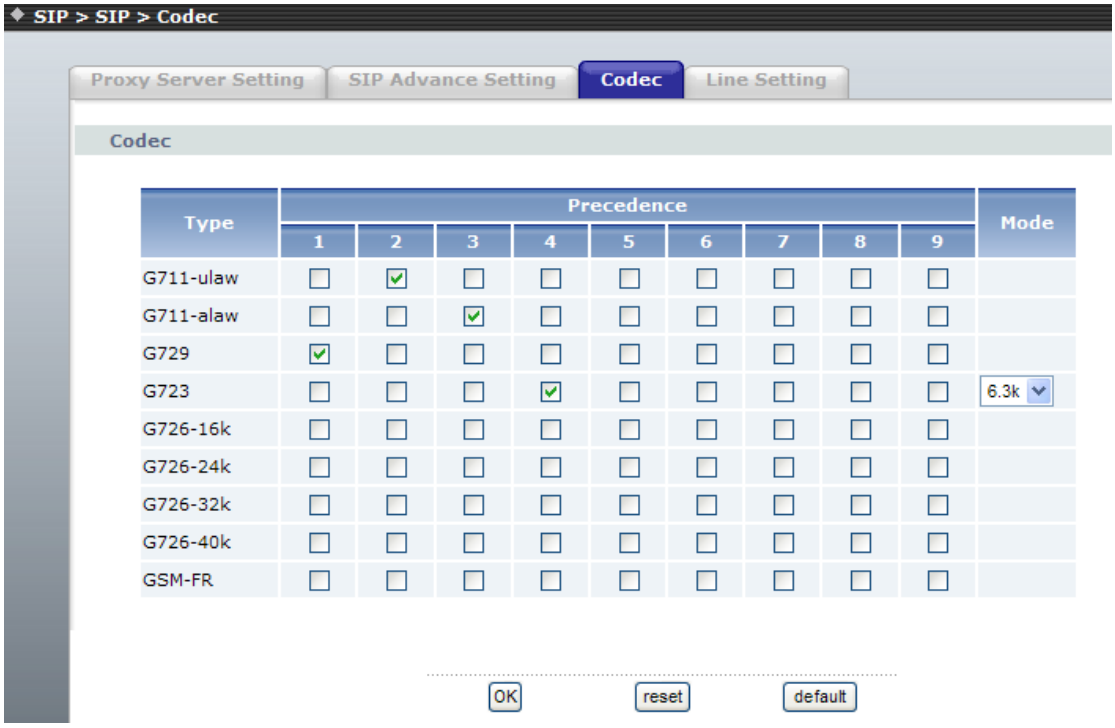
□SIP Account Pooling: SIP account sharing

□Local Port call directly

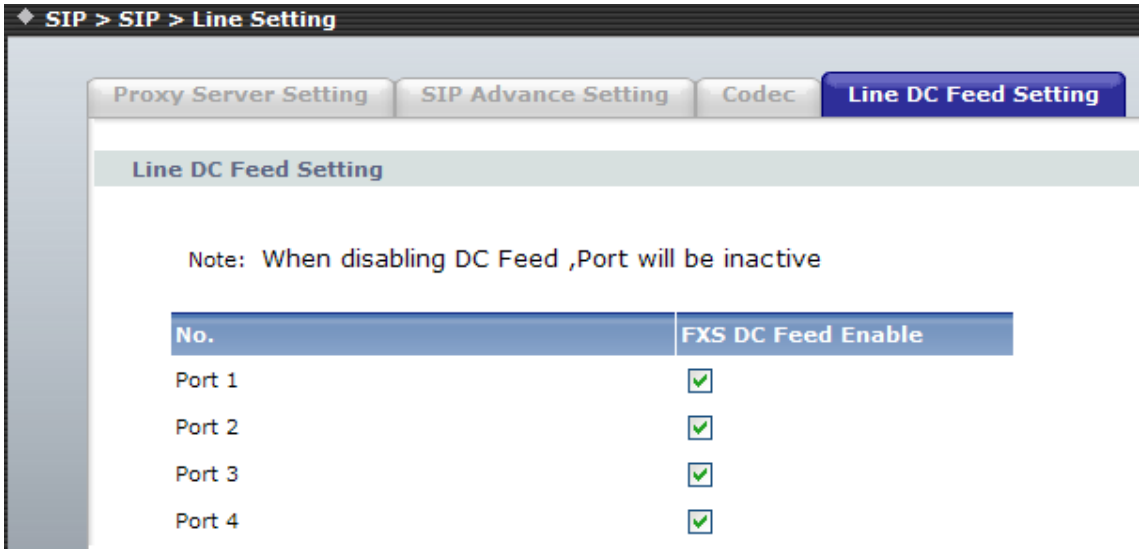
□SIP Invite timeout

4.1.3 Codec

A CODEC is an algorithm for taking voice or video and compressing the information. This type of codec combines analog-to-digital conversion and digital-to-analog conversion functions in a single chip. The Codec is used to compress the voice signal into data packets. Each Codec has different bandwidth requirement. There are 9 kinds of codec, G.711/Ulaw, G.711/Alaw, G.729, G.723(5.3k / 6.3k bps), G.726(16K bps), G.726(24K bps), G.726(32K bps), G.726(40K bps), GSM-FR



4.1.4 Line DC Feed Setting



Port 1: on /off line 1 function.

Port 2: on /off line 2 function.

Port 3: on /off line 3 function.

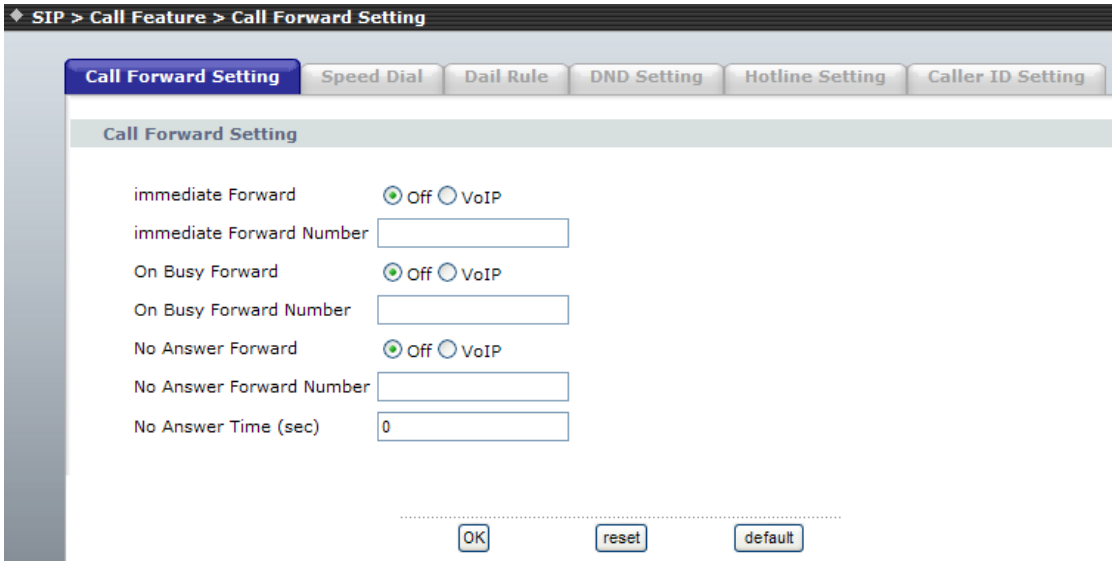
Port 4: on /off line 4 function.

4.2 Telephone function

4.2.1 Call Forward Setting

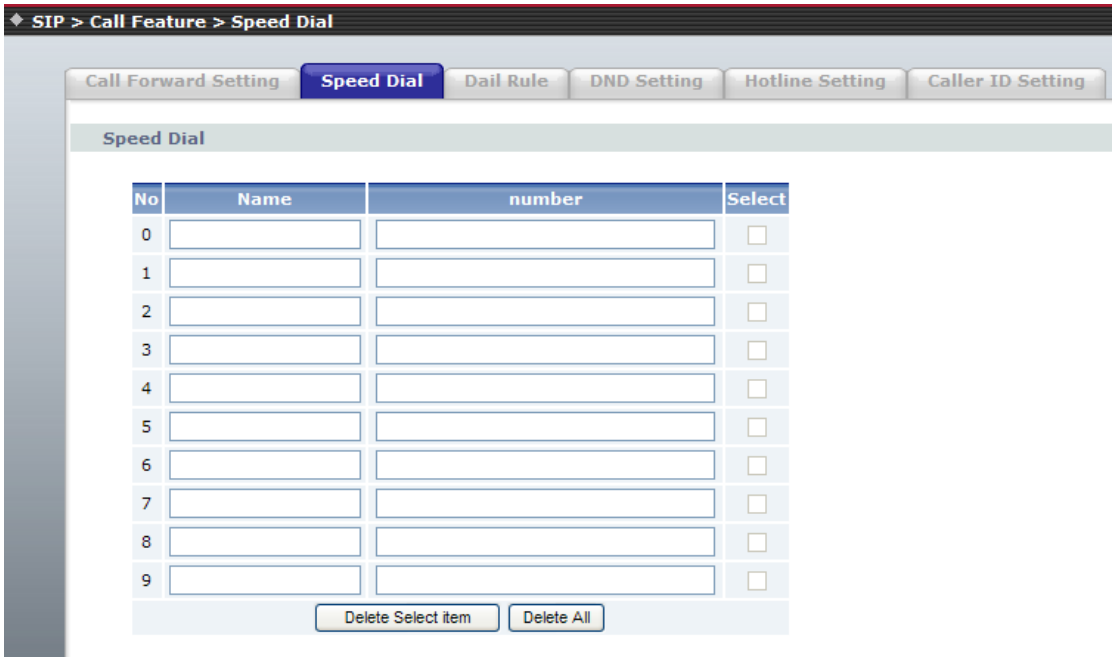
You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose **Immediate Forward, Busy Forward, and No Answer Forward** by click the icon.

Immediate Forward: All incoming call will forward to the number you choose. You can input the phone number. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.



4.2.2 Speed Dial

Speed Dial lets you define a button or a set of buttons to link to a specific number defined in Speed Dial list.

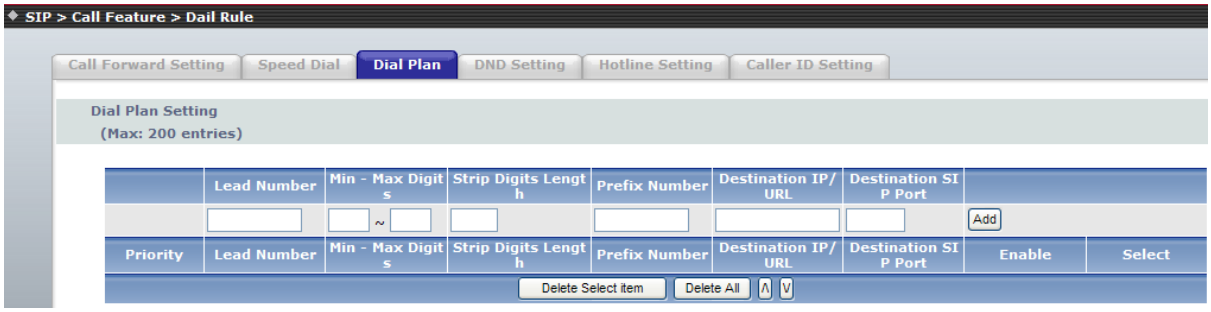


□Name: Phone number note.

□Phone Number: Enter the international number to dial.

4.2.3 Dialing Plan

The “Dialing plan” needs to setup when the users use the method of Peer-to-Peer SIP VoIP call or SIP Proxy Server Mode. The SIP Dialing Plan has two kinds of directions: Outgoing (call out).



□“Lead Number” is the leading digits of the call out dialing number.

□“Min-Max Digits” has two text fields need filled: “Min Length” and “Max Length” is the min/max allowed length you can dial.

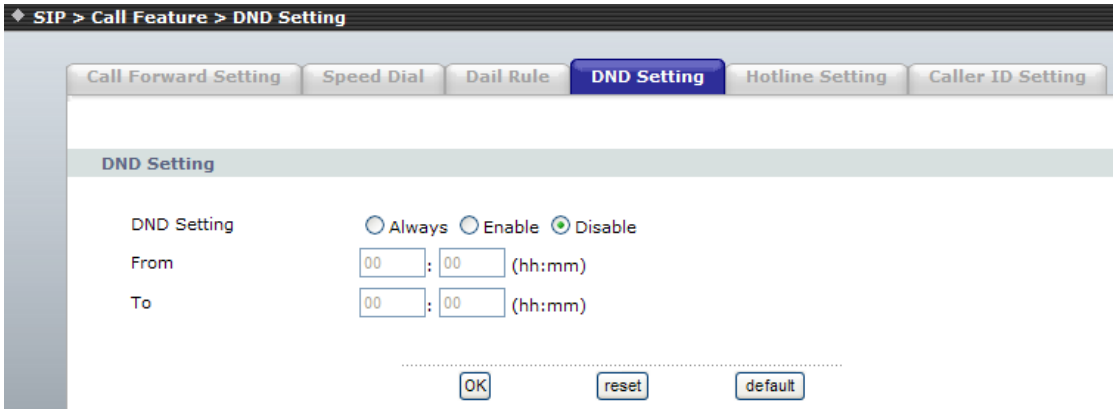
□“Strip Digits Length” is the number of digits that will be stripped from beginning of the dialed number.

□“Prefix Number” is the digits that will be added to the beginning of the dialed number.

□“Destination IP/ URL” is the IP address / Domain Name of the destination TG1600/TG2400/TG3200 series voip gateway that owns this phone number.

□“Destination SIP Port” is port of the destination TG1600/TG2400/TG3200 series voip gateway use.(Default is 5060)

4.2.4 Do Not Disturb



DND Setting: you can setup the DND Setting to keep the device silence. You can choose Always or Enable or Disable.

DND Always: All incoming call will be blocked until disable this feature.

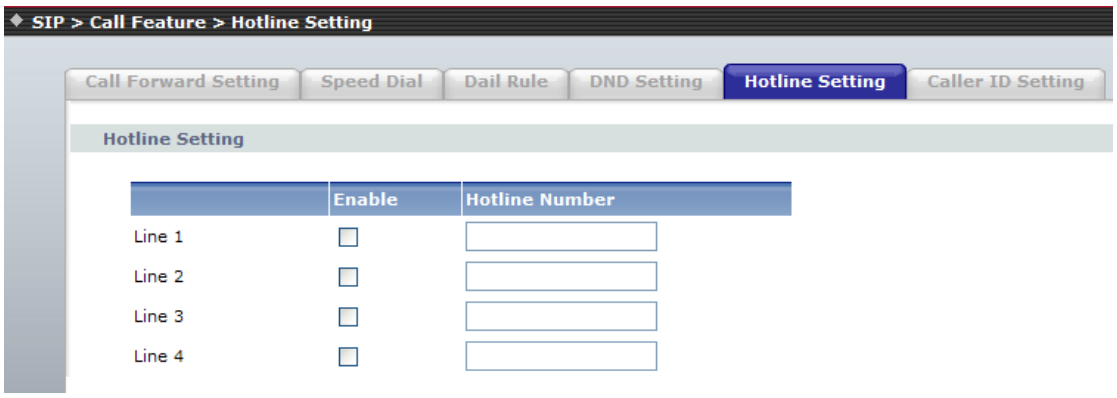
DND Enable: Set Enable and the device will be blocked during the time period. If the “From” time is large than the “To” time, the Block time will from 00:00 to 23:59

When you finished the setting, please click the Submit button.

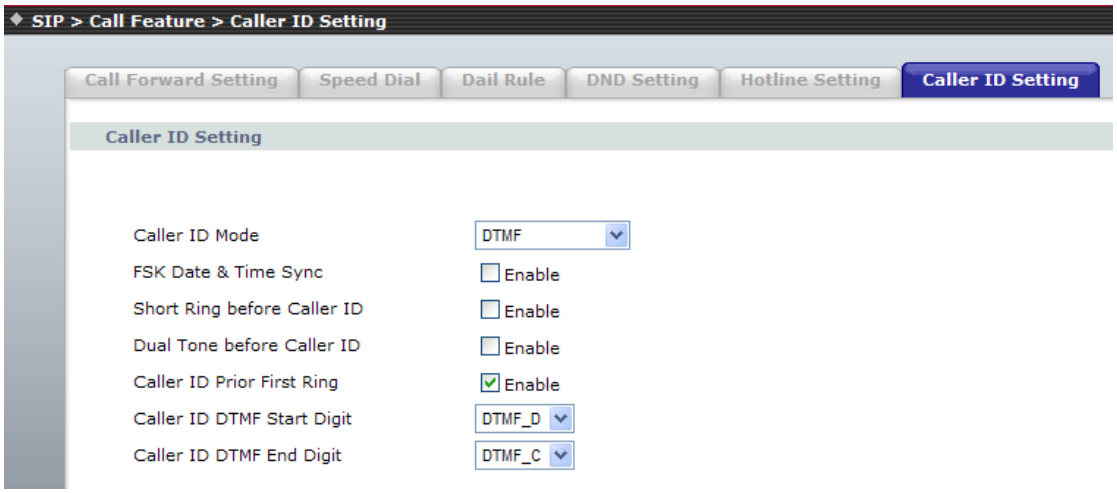
If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will reboot automatically.

4.2.5 Hotline Setting

This function allows you to make a call to a pre-programmed number by only lifting the handset.



4.2.6 Caller ID display



Caller ID mode: support FSK_Bellcore, FSK_ETSI, FSK_BT, DTMF

FSK Date and time synchronization: Enable FSK single sending date and time to show on phone screen

FSK Date & Time Sync: Send FSK Date & Time to display device.

Reverse Polarity before Caller ID: Send Reverse Polarity before Caller ID

Short Ring before Caller ID: Send short ring before caller ID.

Dual Tone before Caller ID: Send Dual Tone before Caller ID.

Caller ID prior First Ring: Send Caller ID before first ring.

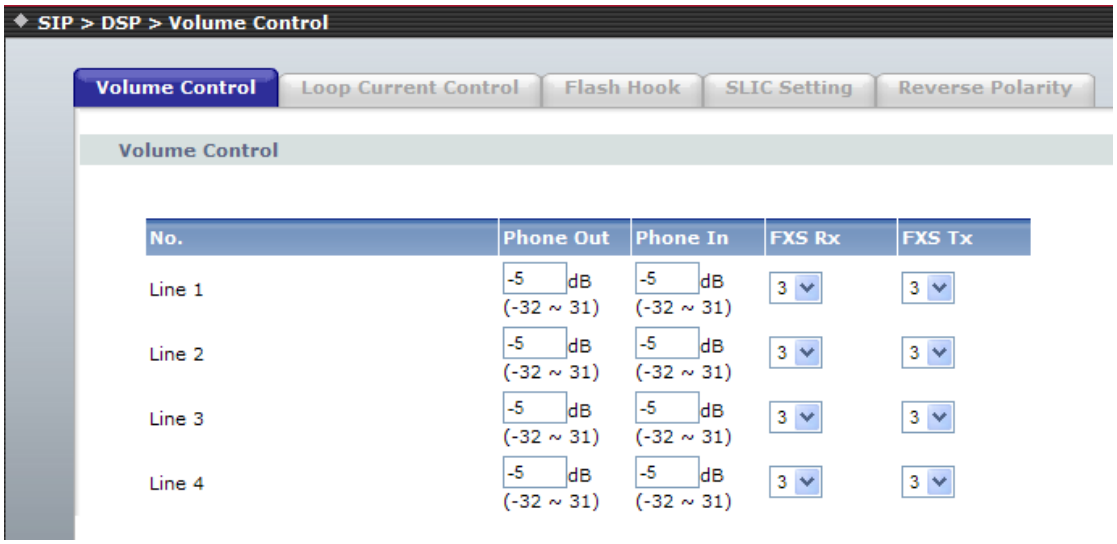
Caller ID DTMF Start Digit: Set Caller ID DTMF start digit.

Caller ID DTMF END Digit: Set Caller ID DTMF start digit.

4.3 DSP

4.3.1 Volume Control

Adjust the volume via each port. There are 2 chips have volume adjust function in GW, one is DSP, another is SLIC

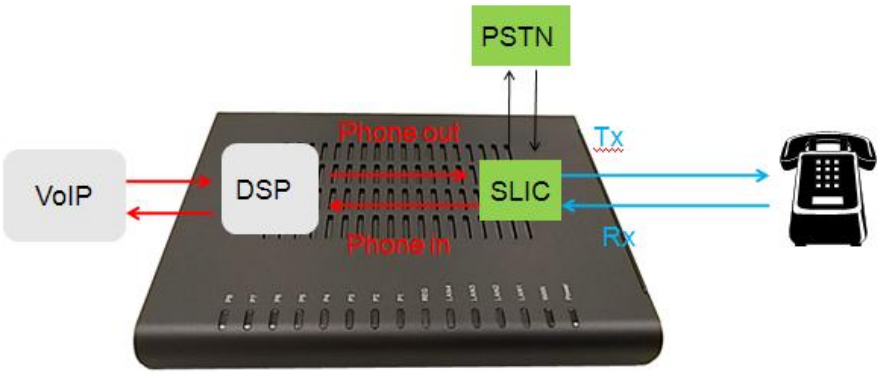


□Phone Out：You may adjust DSP chip to control out put volume, rang from -32~31dB

□Phone In：You may adjust DSP chip to control in put volume , range from -32~31dB

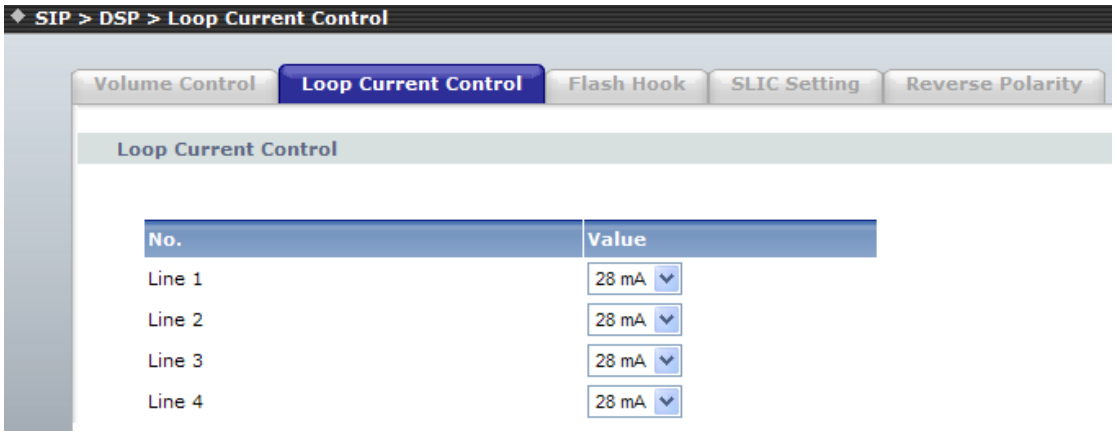
□FXS Tx：YOU may adjust SLIC chip to control out put volume, range from 1~ 9

□FXS Rx：You may adjust SLIC chip to control the in put volume, range from 1~9



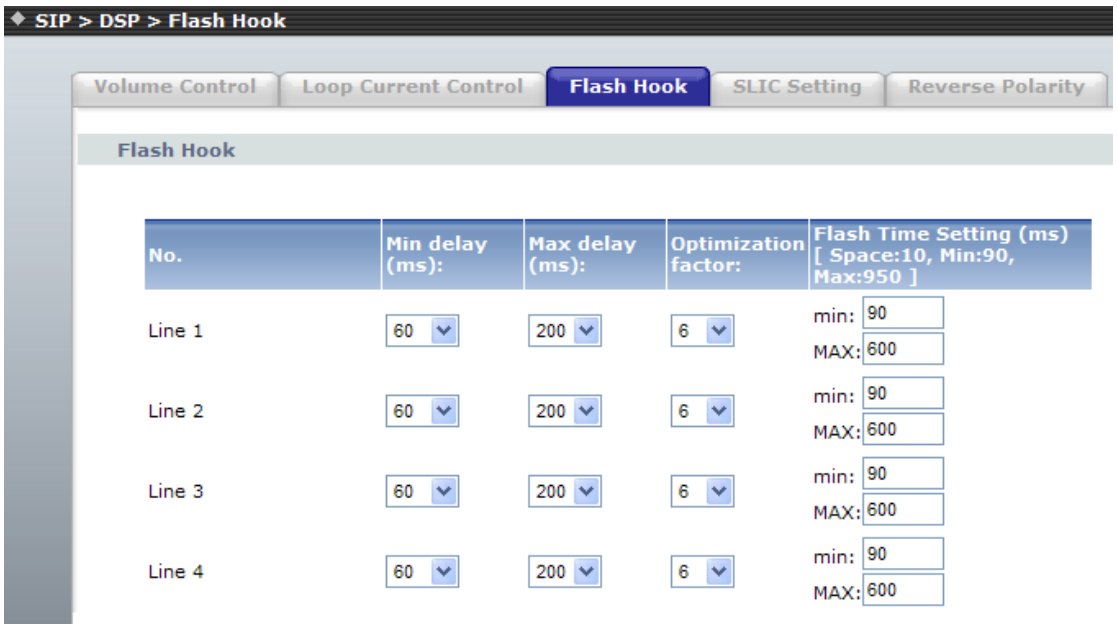
4.3.2 Current control loop

Telephone interface for each group can do loop current control, and adjust your current control for PBX or phone



4.3.3 Flash Hook

The jitter buffer control is a shared data area where voice packets can be stored, collected, and sent to the voice shared Buffer in evenly spaced intervals. Modify in packets arrival time, called jitter, can occur because of network congestion, timing drift, or route changes.



□Min delay (ms): Select min delay buffer time.(40ms – 100 ms)

□Max delay (ms): Select Max delay Buffer time.(130ms – 300ms)

□Optimization factor: Controls quickly the length of the Jitter Buffer is increased when Voice RTP on the network. Default is 6

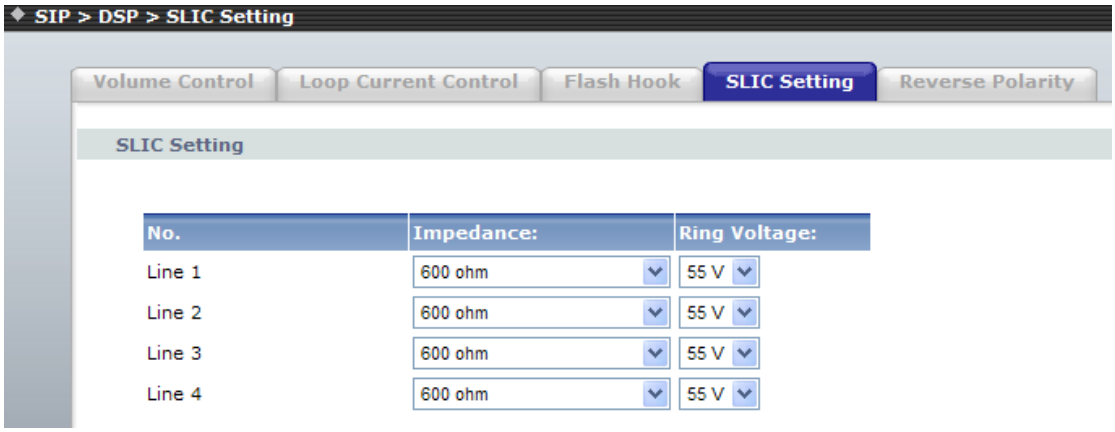
□Flash time Setting: Set Min / max storage Flash Time Setting

4.3.4 SLIC Setting

The phone features adjustable impedance and voltage magnitude, in response to different countries in the phone system, the internal impedance of the phone in some countries there will be slight differences, may cause the phone not working properly, you may adjust the

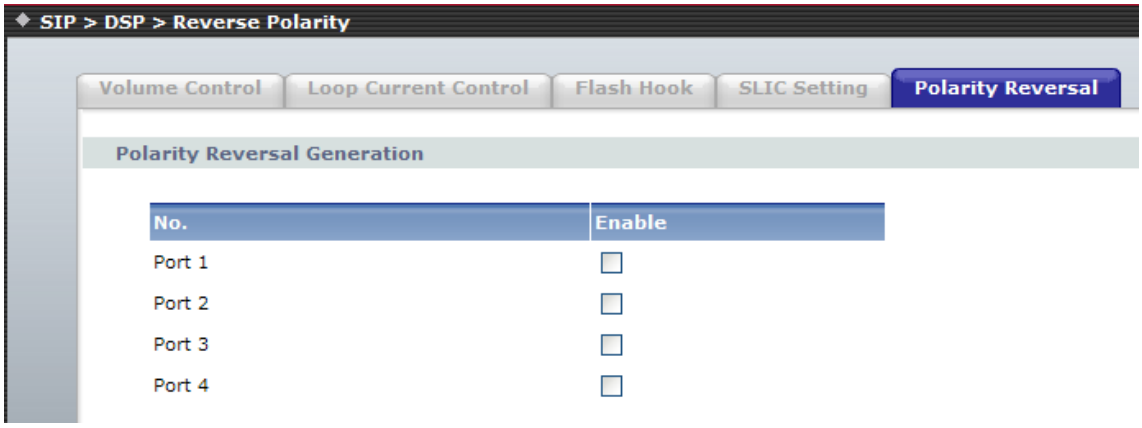
internal Impedance for your phone.

Ring Voltage: Can adjust your phone ring voltage, when your phone rings are not normal, try to adjust this setting



4.3.5 Polarity Reversal

As Callee Answer - Check this box to generate line polarity reversal while the remote user picks up the phone call. As Callee On-Hook - Check this box to generate line polarity reversal while the remote user hangs off the phone.



5

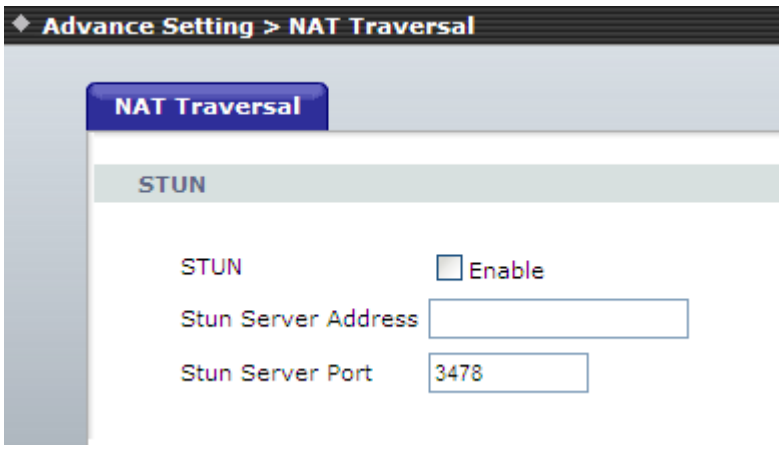
Advance setting

5.1 NAT Traversal

STUN {Simple Traversal of UDP through NATs (Network Address Translation)} is a protocol for assisting devices behind a NAT firewall or router with their packet routing.

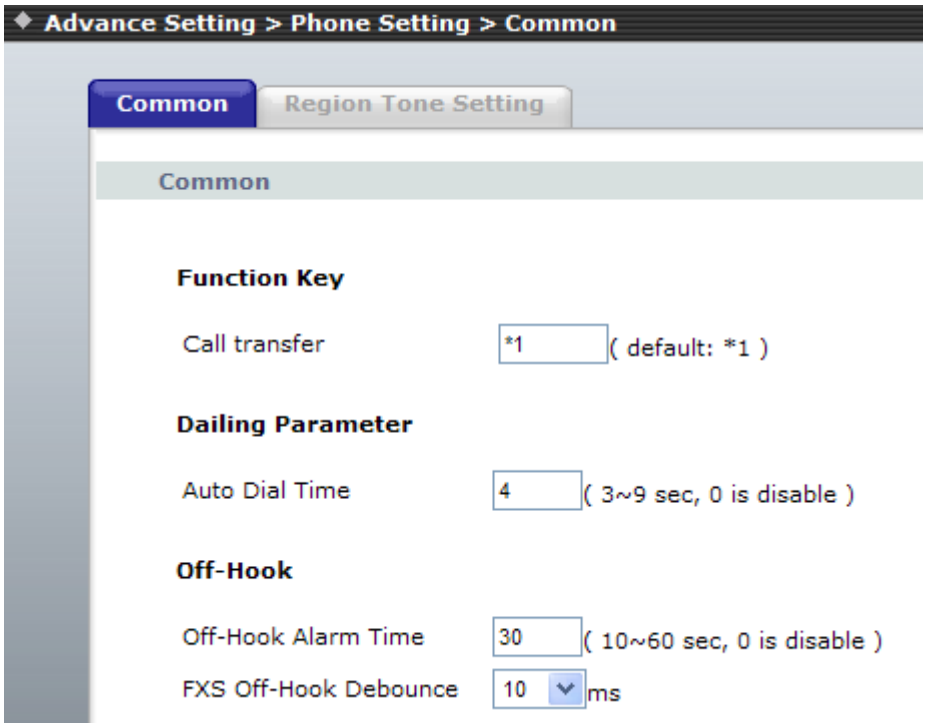
STUN enables a device to find out its public IP address and the type of NAT service its sitting behind.

When you enable the STUN function, you must input the STUN server address.



5.2 Phone setting

5.2.1 Common



□Call transfer: Set your call control keys, it starts to call transfer when phone hang up, default is "* 1"

Dialing Parameter

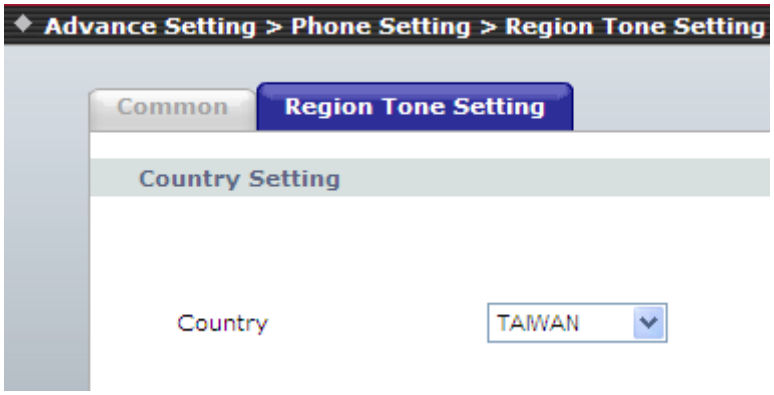
□Auto Dial Time: If no other number is being dialed within this interval, the gateway will terminate this call. Assign the time interval from 1 to 9 seconds.

□Off-Hook Alarm Time: Set has been off-hook, after this time, user will hear alarm.

□FXS off-Hook Debounce: For example, if the value is 10ms, user needs to push the flash key longer than 10ms to hang up the call. For any reason, user may dial the flash key wrongly, if user push the flash key under 10ms, the device will not hang up the call.

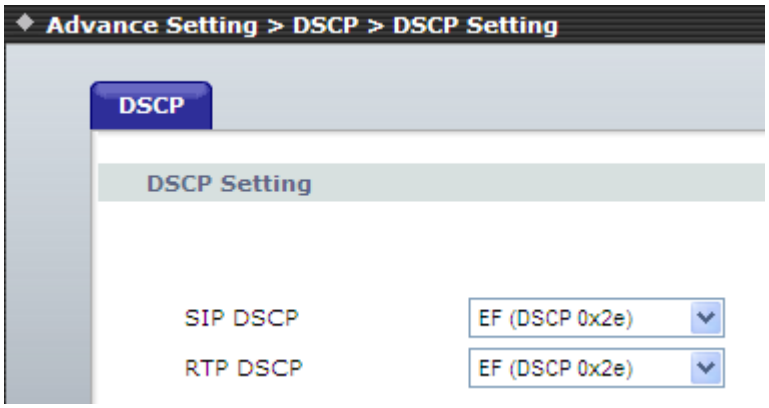
5.2.2 Region Tone Setting

Adjust the tone frequency by country. Select a country from the pull-down menu.



5.3 DSCP Setting

Select the SIP and RTP to adjust the transmission priority of voice and data.

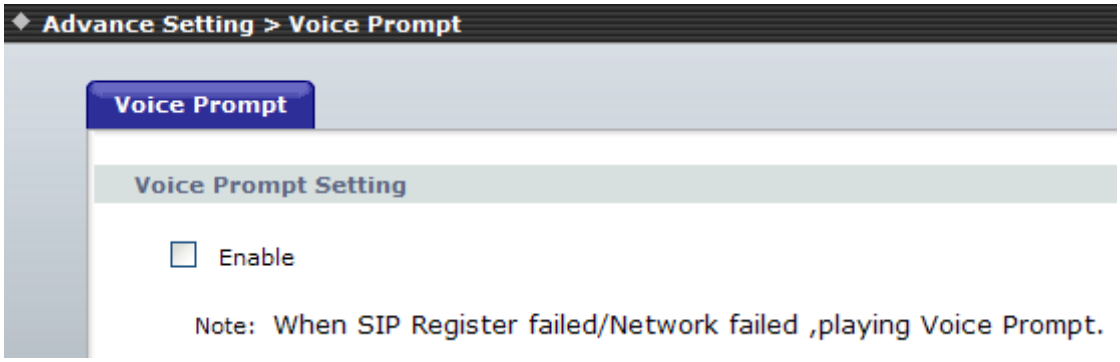


□SIP DSCP: enter the priority for SIP voice transmission. The device creates type of service priority tags; with this priority to voice traffic that is transmits.

□RTP DSCP: enter the priority for RTP voice transmission. The device creates type of service priority tags; with this priority to RTP traffic that is transmits

5.4 Voice Prompt

When SIP register failed or Network failed, playing the Voice Prompt.


Advance Setting > Voice Prompt
Voice Prompt
Voice Prompt Setting
Enable
Note: When SIP Register failed/Network failed ,playing Voice Prompt.

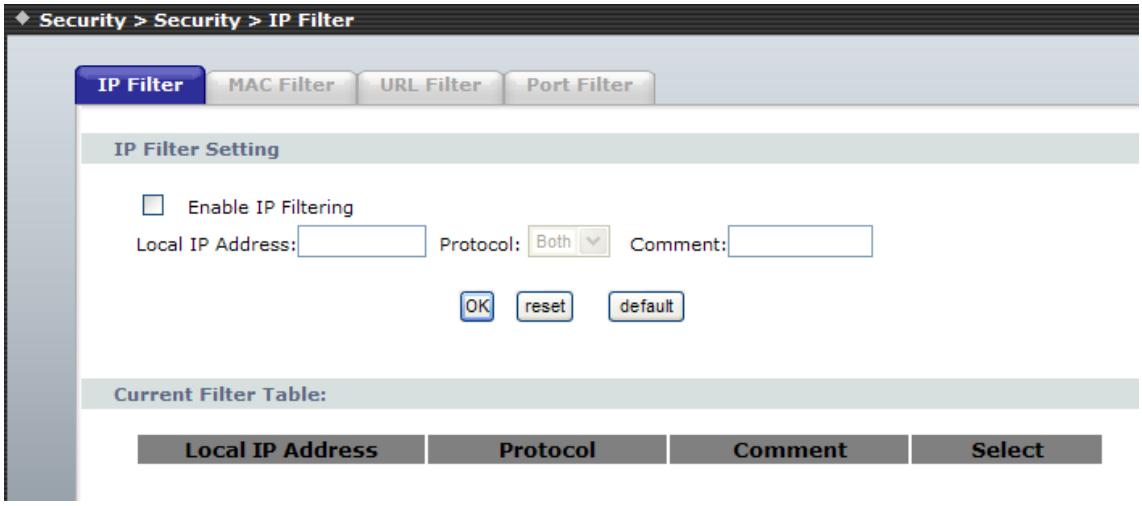
6

Security

6.1 Firewall

6.1.1 IP Filter

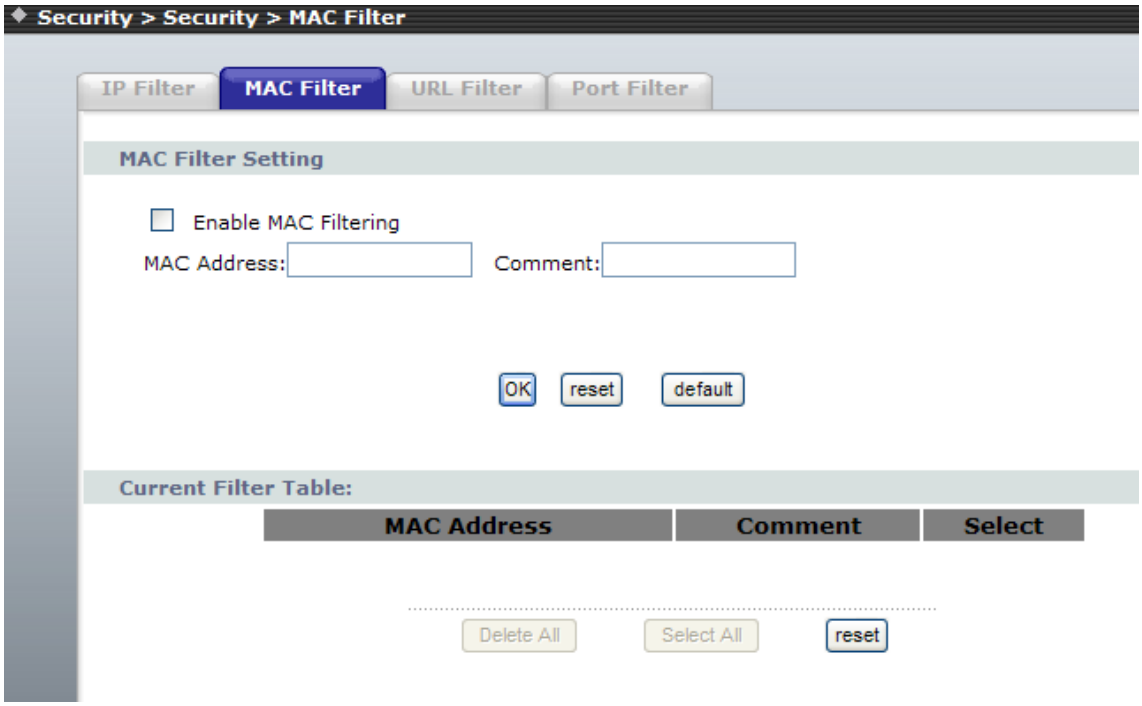
Entries in this table are used to restrict certain types of data packets from your local network to Internet through the Gateway. Use of such filters can be helpful in securing or restricting your local network.



□ Enable/Disable IP Filtering: The IP address filter function, control a network IP address from your local network to Internet through the Gateway, default setting is disable.

6.1.2 Mac Filter

Entries in this table are used to restrict certain types of data packets from your local network to Internet through the Gateway. Use of such filters can be helpful in securing or restricting your local network.

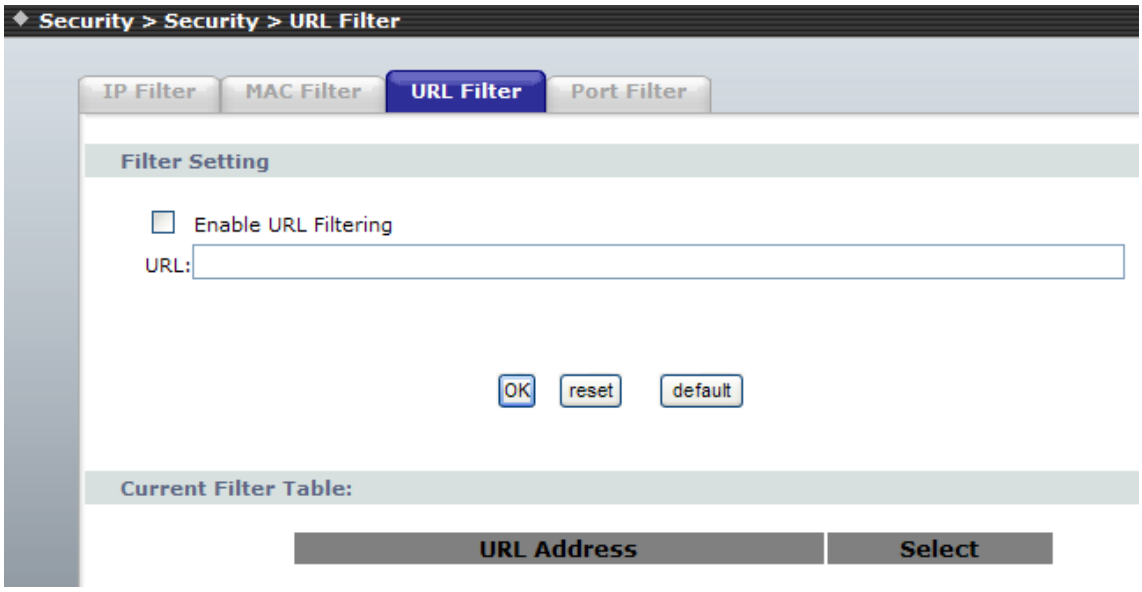


□MAC Enable/Disable: from your local network to Internet through the Gateway MAC filter function, default setting is disable.

6.1.3 URL Filter

URL filter is used to deny LAN users from accessing the internet. Block those URLs which contain keywords listed below.

URL filter allows you to block sites based on a black list and white list. Sites matching the black list but not matching the white list will be automatically blocked and closed.



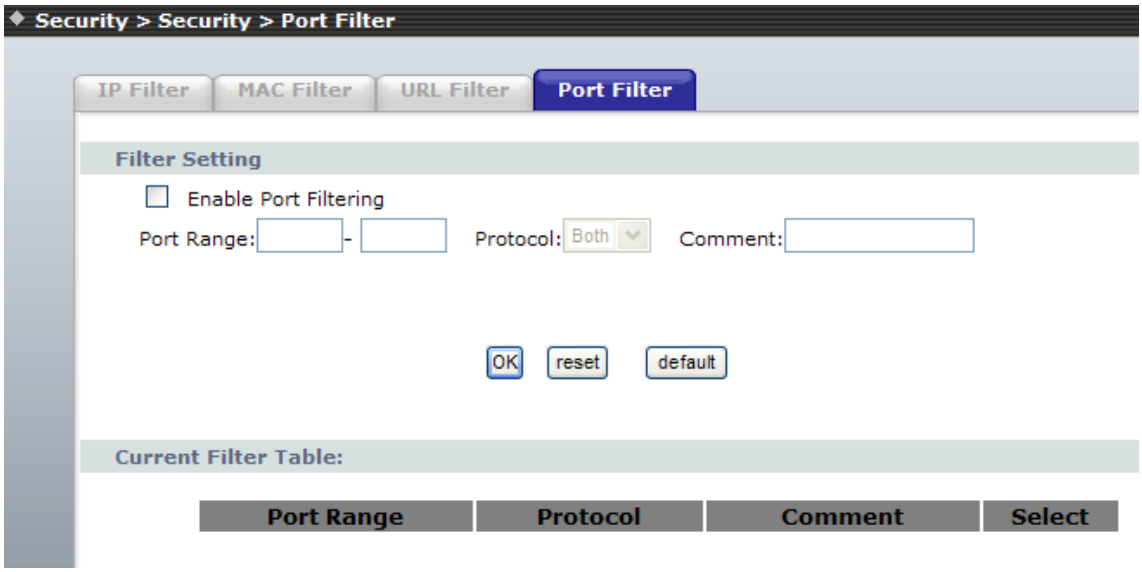
□Enable: Enable/Disable the URL filter function, default setting is Disable.

□URL Address: Enter the filter URL.

Example: “http://www.yahoo.com/”

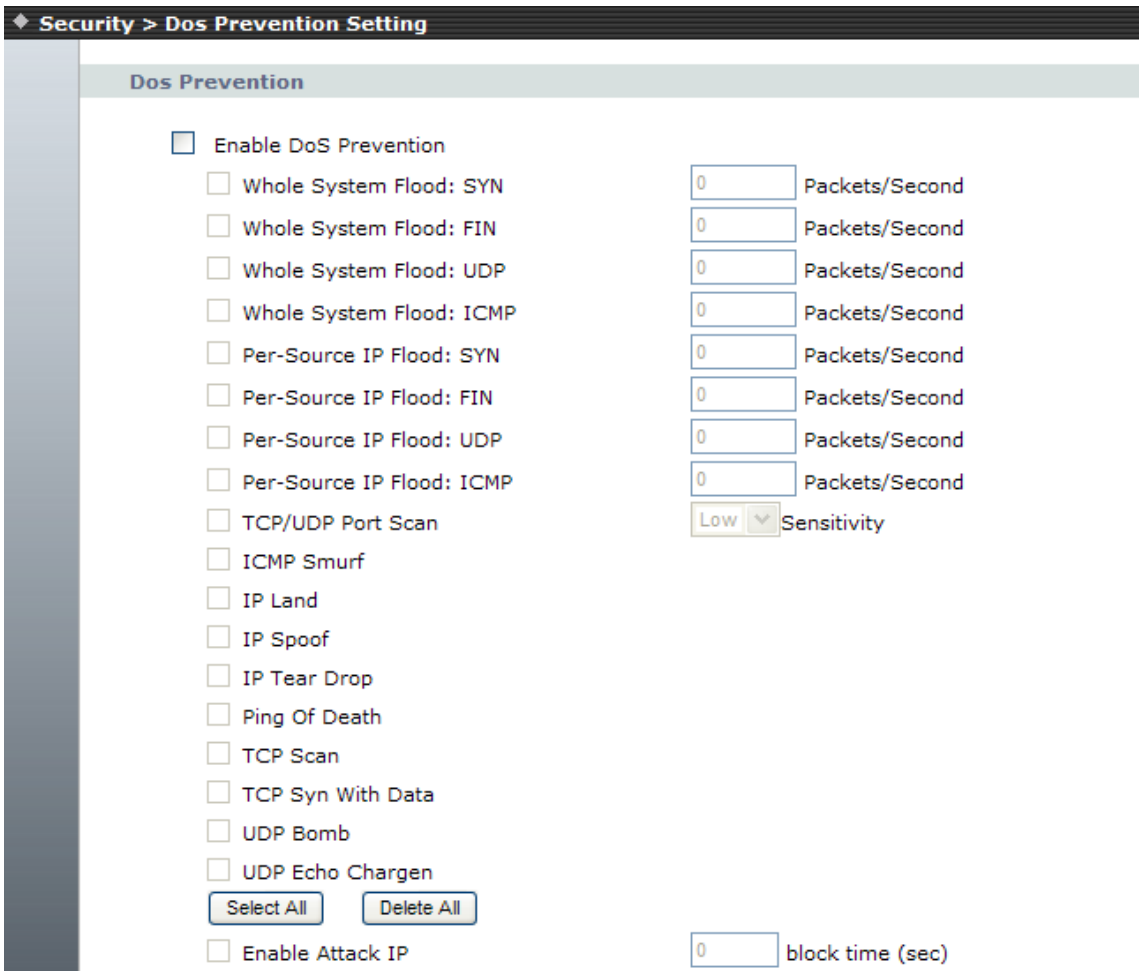
6.1.4 Port Filter

Entries in this table are used to restrict certain types of data packets from your local network to Internet through the Gateway. Use of such filters can be helpful in securing or restricting your local network



6.2 DOS Prevention Setting

Against distributed denial of service attack (DoS) attacks, turn on this feature can be stopped to prevent problems resulting from access to certain resource, you can check the options related to denial of service


Security > Dos Prevention Setting
Dos Prevention
Enable DoS Prevention
Whole System Flood: SYN
0
Packets/Second
Whole System Flood: FIN
0
Packets/Second
Whole System Flood: UDP
0
Packets/Second
Whole System Flood: ICMP
0
Packets/Second
Per-Source IP Flood: SYN
0
Packets/Second
Per-Source IP Flood: FIN
0
Packets/Second
Per-Source IP Flood: UDP
0
Packets/Second
Per-Source IP Flood: ICMP
0
Packets/Second
TCP/UDP Port Scan
Low
Sensitivity
ICMP Smurf
IP Land
IP Spoof
IP Tear Drop
Ping Of Death
TCP Scan
TCP Syn With Data
UDP Bomb
UDP Echo Chargen
Select All
Delete All
Enable Attack IP
0
block time (sec)

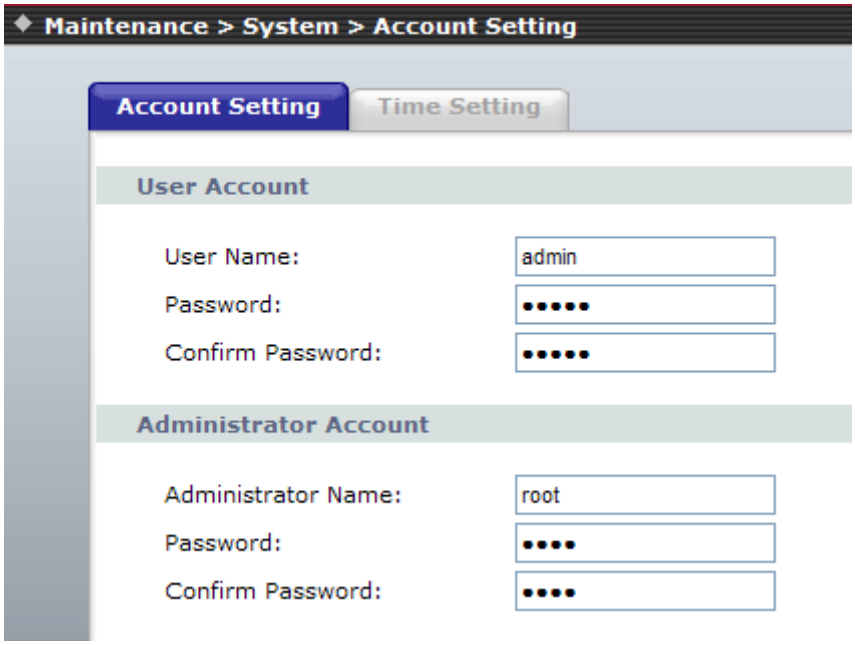
7

Maintenance

7.1 System

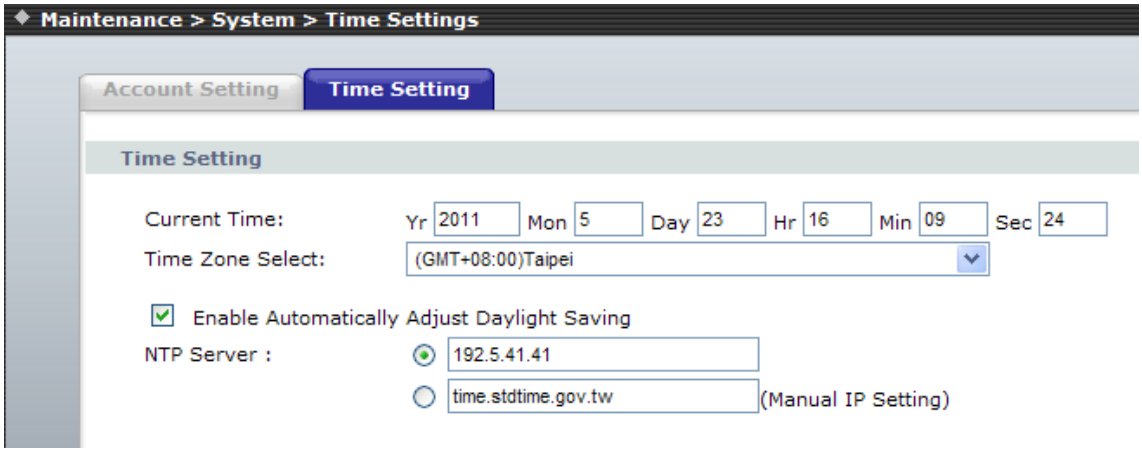
7.1.1 Account Setting

Users can log in via the web interface by entering the username and password to view administration pages. Only the administrator account has permissions to change user account password



7.1.2 Time Setting

The TG1600/TG2400/TG3200 series voip gateway set the date and time by using the Time Protocol (SNTP). You can manually set the timer, or enable automatic time server settings to update the time through the time server.



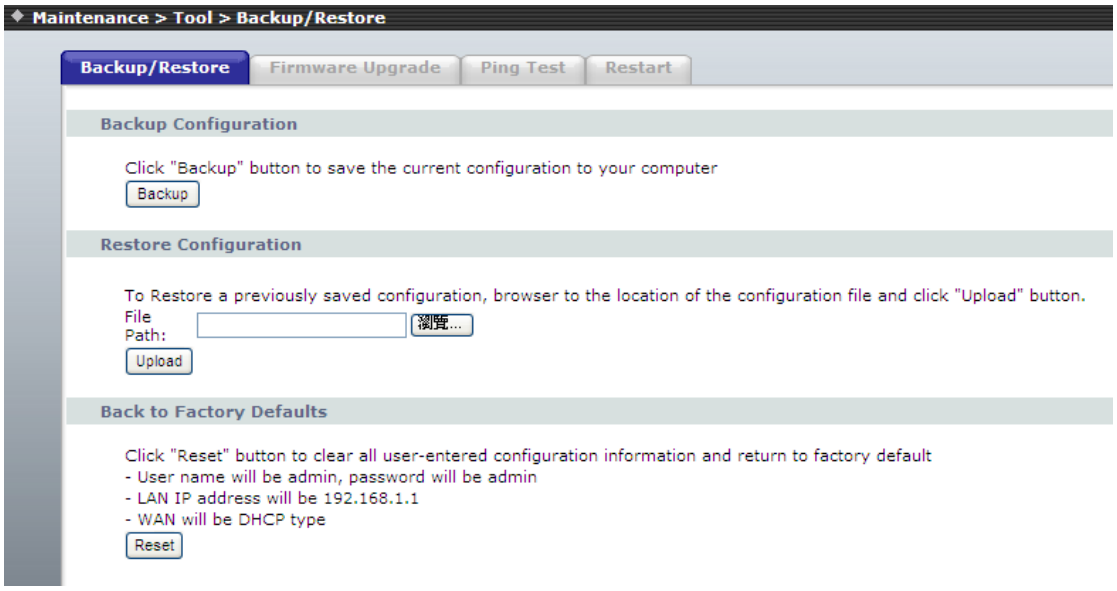
□Time Zone Select: Choose your time zone

□NTP Server: Select NTP server.

7.2 Tool

7.2.1 Backup / Restore

This page allows you save current settings to a file or reload the settings from the file which was saved previously. Besides, you could reset the current configuration to factory default.


Maintenance > Tool > Backup/Restore
Backup/Restore
Firmware Upgrade
Ping Test
Restart
Backup Configuration
Click "Backup" button to save the current configuration to your computer
Backup
Restore Configuration
To Restore a previously saved configuration, browser to the location of the configuration file and click "Upload" button.
File Path:
瀏覽...
Upload
Back to Factory Defaults
Click "Reset" button to clear all user-entered configuration information and return to factory default
- User name will be admin, password will be admin
- LAN IP address will be 192.168.1.1
- WAN will be DHCP type
Reset

□Backup: click “Backup” button to save the current configuration of your system to your computer.

□Restore: To restore a previously saved configuration file to your system, “browser” to the location of the configuration file and click “Upload”

□Back to Factory Defaults: click “Default” button to clear all user-entered configuration information and return to factory defaults. After resetting, the

-Password will be admin

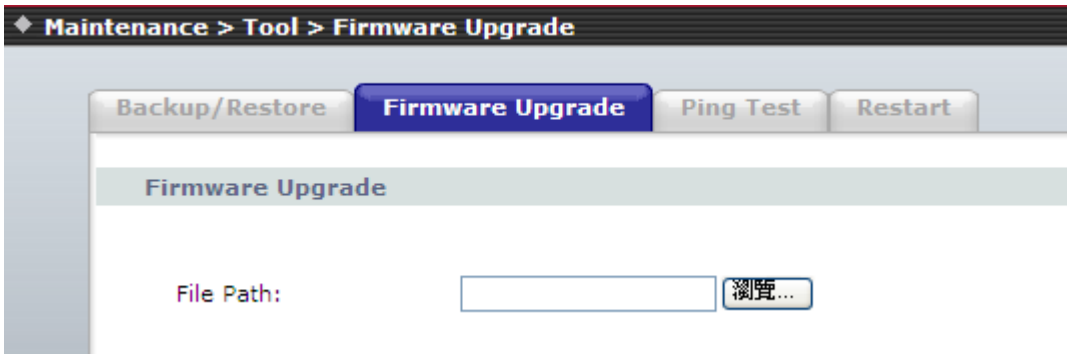
-Lan IP Address: 192.168.1.1

-DHCP will be reset to server

7.2.2 Firmware Upgrade

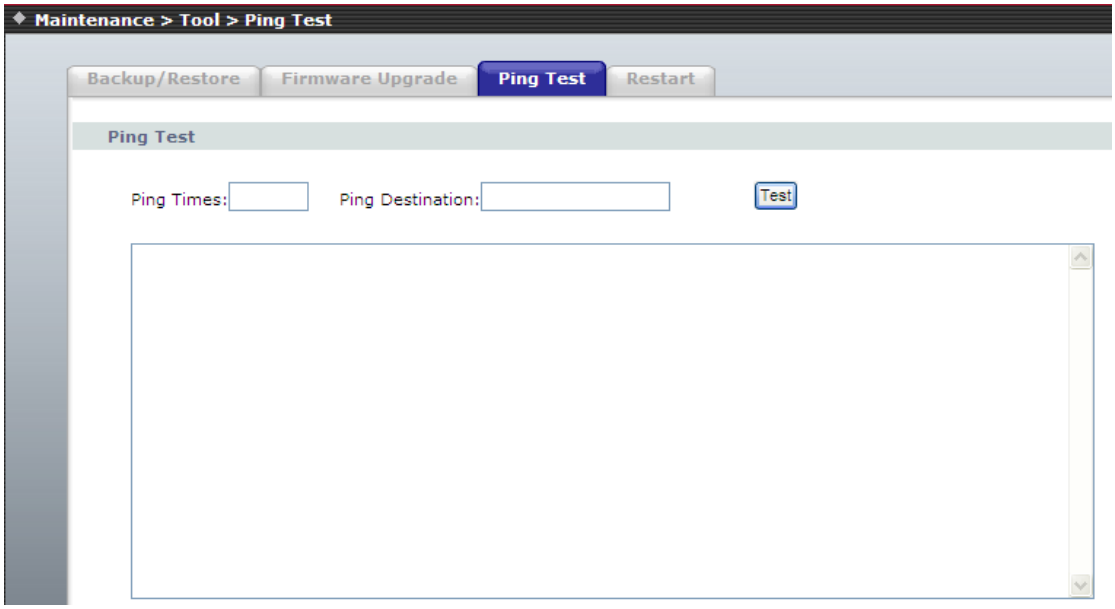
This page allows you upgrade the Access Point firmware to new version. Please note, do not power off the device during the upload because it may crash the system.

Notice: It won't change the system parameter you set when upgrading firmware, but better you save the system setting before upgrade firmware.



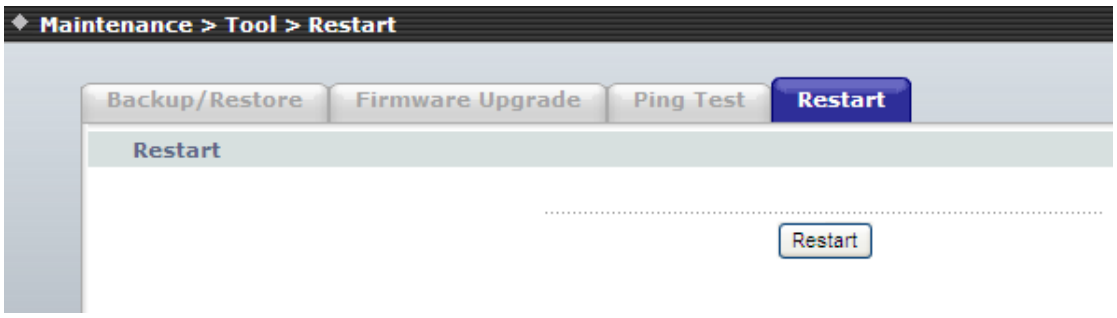
7.2.3 Ping testing

Network traffic diagnostic tools can be used to check the TG1600/TG2400/TG3200 series voip gateway is connected to the Internet. It will send ping packets to a remote target location of a particular host, and test whether the link. Enter the address in the target host name or IP address (for example: www.ippbx.com or 216.115.108.245), then click the "test" button.



7.2.4 Restart

Click “Restart” button to restart the device. For some reasons, the device is not responding correctly, if you want to restart the TG1600/TG2400/TG3200 series voip gateway, you can just click the Restart button, than the VoIP equipment will reboot automatically.



8

Installation structure

8.1 Installation structure

(FXS Gateway)